Cisco AS5300/Voice Gateway

A NATURAL EXTENSION OF CISCO SYSTEMS IP NETWORKING EXPERTISE, THE CISCO AS5300/VOICE GATEWAY RELAYS HIGH QUALITY VOICE AND FAX TRAFFIC ACROSS AN IP NETWORK.

The Cisco AS5300 is an award-winning dialup remote access server and voice-over-IP (VoIP) gateway. When equipped with voice feature cards (VFCs) and voice-enabled Cisco IOS® software, the AS5300 supports carrier-class VoIP and fax over IP services.

Cisco IOS (Internetworking Operating System) software offers a powerful array of quality-of-service (QoS) mechanisms, variable frame sizing, and standards-based H.323 controls, which provide industry-leading voice quality and call control routing to deliver enhanced services. Mier Communications testing rated the voice-over-IP gateway capabilities of the Cisco AS5300/Voice Gateway as best in the areas of voice quality, latency, and bandwidth requirement.

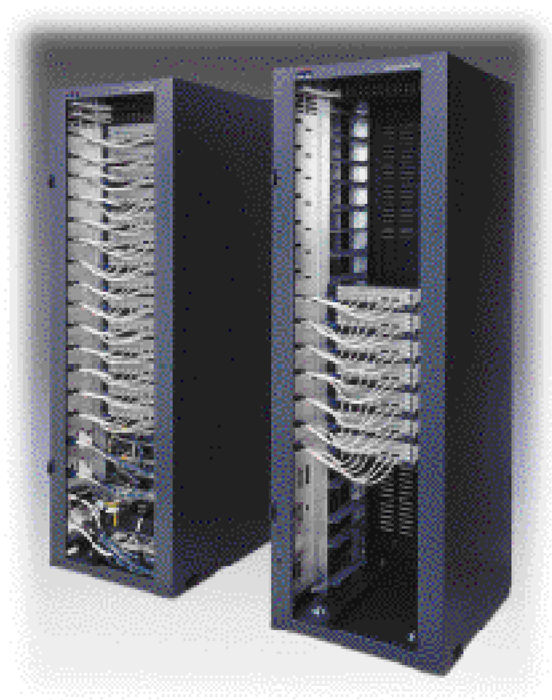
In addition to being H.323 compliant, the Cisco AS5300/Voice Gateway supports a family of industry- standard voice CODECs and provides echo cancellation and voice activity detection (VAD)/silence suppression. It offers an integrated interactive voice response (IVR) application that provides voice prompts and digit collection in order to authenticate the user and identify the call destination. Users can readily interface with Public Switched Telephone Network (PSTN) digital switches or PBXs, and existing RADIUS authentication and billing servers.

The Cisco AS5300 voice/fax feature cards are coprocessor cards, each with a powerful Reduced Instructions Set Computer (RISC) engine and dedicated, high-performance digital signal processors (DSPs) to ensure predictable, real-time voice processing. The design couples this coprocessor with direct access to the Cisco AS5300/Voice Gateway routing engine for streamlined packet forwarding. The Cisco AS5300/Voice Gateway can accept two voice/fax feature cards, so the Cisco AS5300/Voice Gateway can scale up to 96/120 voice connections within a single chassis.

Figure 1 Cisco AS5300/Voice Gateway with High Density Voice/Fax Feature Card for VoIP; Award-Winning, Carrier-Class, and H.323 Compliant

Cisco AS5300/Voice Gateway is ideally suited for a stacked configuration to create a single virtual dial pool for large-scale service provider applications. For example, the award-winning AccessPath™-VS3 is the Cisco preconfigured, pretested stacked solution for VoIP. (For more information, see the AP-VS3 data sheet). Providing the industry's broadest family of compatible products, Cisco enables customers to pick the right starting point for a "pay-as-you-grow" rollout without compromising future capabilities.

Figure 2 AccessPath-VS3, Large Scale Carrier Class VoIP Solution



Quality of Service

Today, major enterprises and service providers are deploying worldwide toll-quality VoIP networks. Cisco voice technology maintains carrier-quality communications in the face of most adverse network conditions, including packet delay and packet loss. Both packet loss and packet delay can have a significant adverse impact on speech quality.

The high-performance voice coprocessor design of Cisco voice gateways minimizes delay and packet loss during the voice encoding and packetization process. Cisco QoS features, including IP Precedence, Resource Reservation Protocol (RSVP), Weighted Fair Queuing (WFQ), Weighted Random Early Detection (WRED), and Multiclass Multilink PPP (MP) fragmentation and interleaving, implemented on both the voice gateways and backbone routing infrastructure, can provide a low-latency, high-reliability path for sensitive voice traffic through today's networks. In Mier Communications tests, the Cisco AS5300/Voice Gateway exhibited the lowest latency of any VoIP product—using industry-standard H.323 and G.729 CODEC. The Cisco AS5300/Voice Gateway VoIP solution typical latency clocked in their lab was only 70 milliseconds.

Applications

Service Provider Long-Distance Services

Service providers (SPs) can leverage their existing IP infrastructure to deploy VoIP. The Cisco packet telephony solution is based on H.323, an ITU standard that provides a foundation for data, audio, and video, and communications across IP-based networks. Because SPs already offer Internet access, they can readily offer long-distance service by incrementally adding voice-enabled Cisco AS5300/Voice Gateway ports, additional Primary Rate Interface (PRI), T1, or E1 interfaces to the PSTN, and a gatekeeper to serve multiple gateways. The Cisco 2600 and 3600 can be utilized as a gatekeeper. The Cisco AS5300/Voice Gateway voice gateway is also interoperable with other vendors' H.323 gatekeepers. The service provider can use existing RADIUS servers for authentication, authorization, and accounting (AAA) as well as existing routers and Ethernet switches located in the POPs.

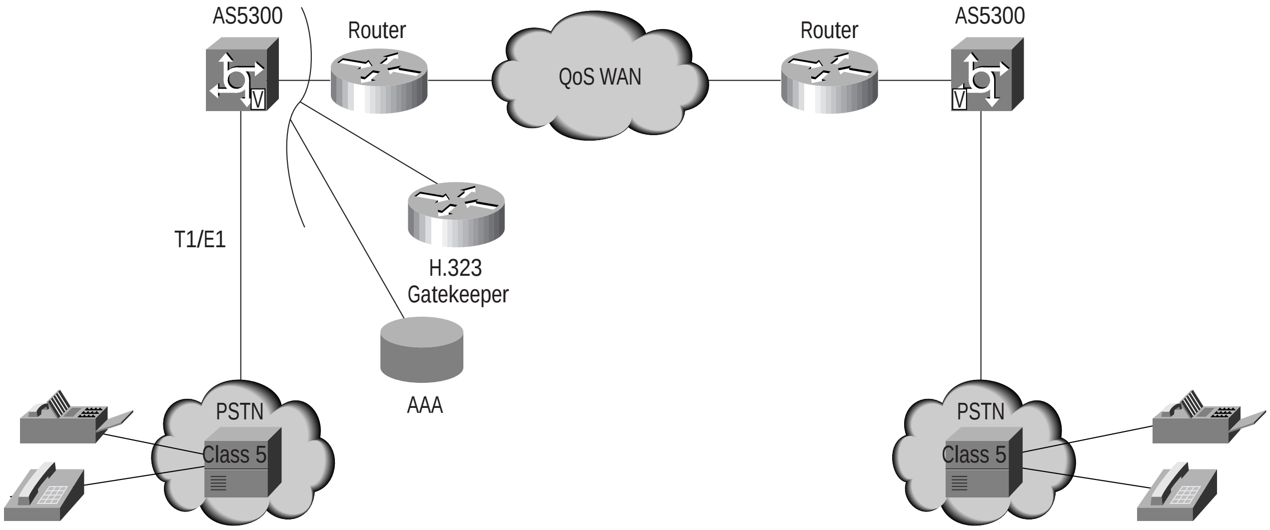
The voice gateway application software enables the router to connect voice calls between PBXs, key systems, or PSTN circuits, transporting the conversations across an IP network. Incoming calls are terminated on the voice/fax feature card, where the voice is encoded using ITU standard algorithms, compressed and encapsulated in Real-Time Protocol (RTP) packets.

Additionally, the Cisco AS5300/Voice Gateway voice gateway has an IVR application that provides voice prompts and digit collection in order to authenticate the user and identify the call destination.

A variety of IVR scripts are provided by the Cisco IOS software; for example:

- Announcement—can be used as part of a script to greet the user and identify the service
- Automatic number identification (ANI)-based automatic authentication

Figure 3 Service Provider Long-Distance Services



- ANI for authentication and dialed-number identification service (DNIS) for call routing
- Account number and password for authentication required
- Fax hop on/off—script that supports fax redialers, which are small boxes connected between a fax machine and the phone line; they store the destination phone number dialed, call the local gateway, and then enter the destination number in response to fax tone prompts; they can also enter an account number if required.

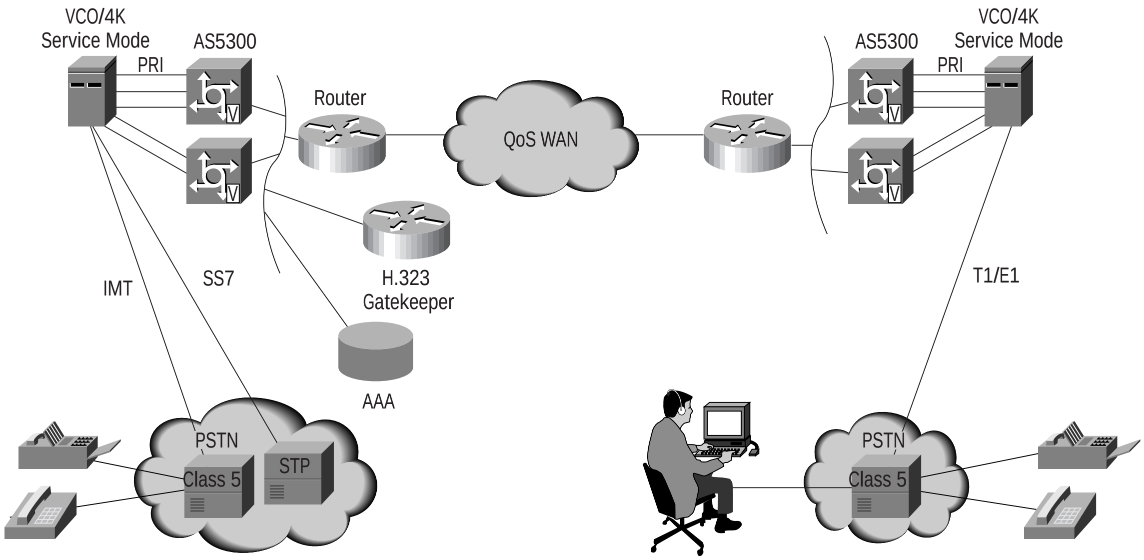
The scripts for these sequences are embedded in the Cisco IOS software. The actual audio prompts are stored as.au files and can be modified by the service provider. For example, the SP might want to include its company name in the prompt message. The prompts can be recorded on a PC and downloaded to the gateway.

The Cisco IOS software also provides several commands for such IVR-related tasks as replacing an audio prompt file, specifying when to use particular IVR scripts, and listing the available IVR scripts.

With this equipment in Figure 3, SPs can carry voice traffic over packet networks and thereby:

- Provide new services beyond basic Internet access
- Offer competitively priced voice services by utilizing their lower-cost IP infrastructure
- Expand their customer base to millions of customers
- Increase revenue from existing POPs
- Differentiate by bundling voice and data services
- Offer business-managed voice and data services

Figure 4   SS7 Enabled Long Distance and Enhanced Services Including Unified Messaging, Voicemail, Calling Card, Customized Billing, and Internet Call Waiting



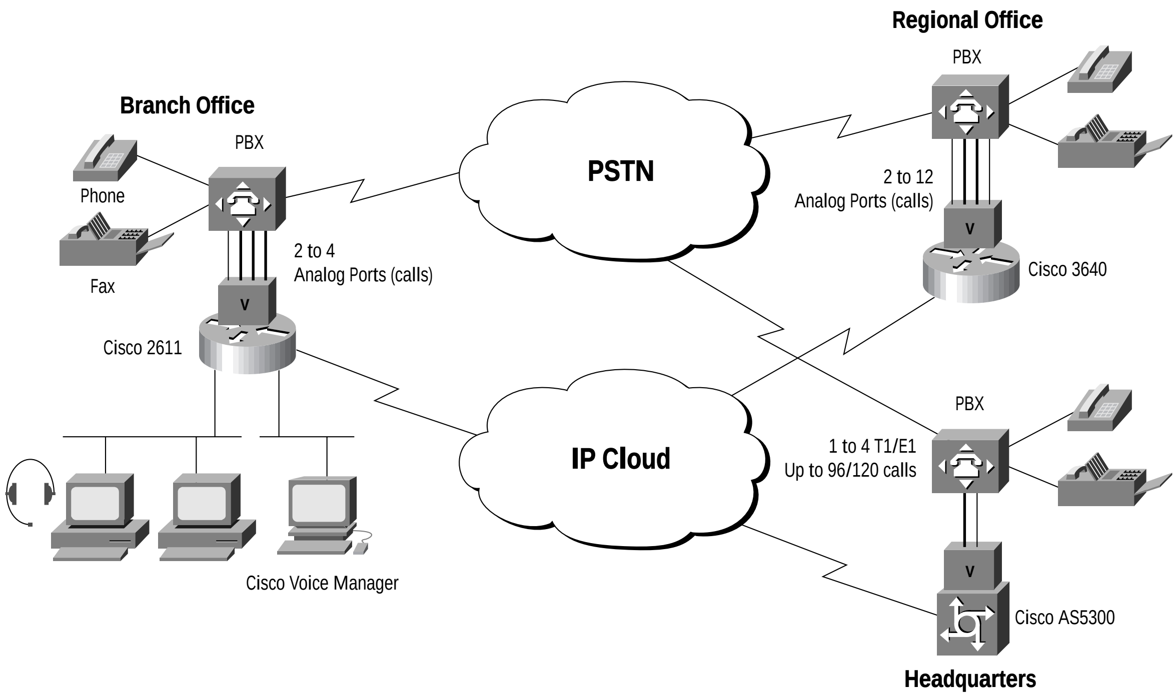
Advanced Service Provider Long Distance Services

Service Providers can further enhance their VoIP service offerings by adding a VCO/4K programmable switch (Figure 4). The VCO/4K enables more advanced voice services such as voice/fax messaging, intelligent 800- number routing, voice-activated dialing, calling card, personal-assistant services, and Web-based conferencing. In addition, it can integrate seamlessly into existing telecommunication infrastructures with application- controlled Signaling System 7 (SS7) signaling to connect to public networks worldwide. VCO/4K performs translation of SS7 ISUP, and Transaction capabilities application part (TCAP) to PRI, so service providers can integrate a broad range of service switching point (SSP) capabilities on new and existing VCO/4K switches at a fraction of the cost of an access tandem switch.

With the advanced network in place, a service provider can offer unified messaging, Internet call waiting (alerting users to incoming voice calls while on-line) and virtual second line (ability to make and receive voice calls from the user's PC while on the Internet). These services can greatly increase the Internet SP's (ISP's) revenue streams from Internet access subscribers.

To expand its service coverage, an individual ISP might choose to partner with ISPs in other regions or join a consortium to provide widespread coverage. Settlement firms allow ISPs to provide national coverage by exchanging voice traffic with other ISPs. Cisco has also introduced support for Open Settlements Protocol, a standard being developed to facilitate the exchange of VoIP traffic.

Figure 5   Enterprise Intranet Phone Calling and Faxing



Enterprise Intranet Phone Calling and Faxing

A common application for the Cisco AS5300/Voice Gateway cards is intranet phone calling and faxing. Using the Cisco AS5300/Voice Gateway, along with the Cisco 3600 and 2600 with voice modules, companies can significantly reduce their long-distance telephone and fax charges by routing their interoffice voice and fax traffic over their existing IP network (Figure 5). In intranets, administrators can monitor and control service levels and, therefore, achieve and maintain toll-quality voice and fax transmissions on their data networks by using the Cisco packet telephony gateway-enabled family of products.

Because the Cisco AS5300/Voice Gateway, as well as the Cisco 3600 and 2600, all work with standard phone and fax equipment, companies can shift their interoffice voice and fax traffic from their voice network to their data network without needing to retrain users. Transparent to the user, a phone connected through a PBX to a voice port on the Cisco AS5300/Voice Gateway at headquarters can call over a WAN connection to a phone connected to a key system that is connected to a Cisco 3600 at the regional office, or to a phone directly connected to a Cisco 2600 at the branch office. Some smaller offices may not need the full functionality of a PBX, as in this case, where the Cisco 3600 and 2600 are sufficient. Cisco packet telephony products seamlessly route calls or faxes to the IP network or out to the PSTN, depending on settings established by the network administrator. Cost savings from deploying the Cisco voice/fax solution to handle intranet phone calls and faxes will typically cover the upfront equipment investment in a matter of months.

Table 1   Packet Telephony Gateway-Enabled Products from Cisco

Number of Voice Ports	Recommended Cisco Packet Telephony Gateway-Enabled Product
Up to 2520	Cisco AccessPath VS3 digital T1/E1 interfaces
Up to 120	Cisco AS5300 digital T1/E1 interfaces
Up to 12	Cisco 3600 analog/ BRI
Up to 4	Cisco 2600 analog/ BRI

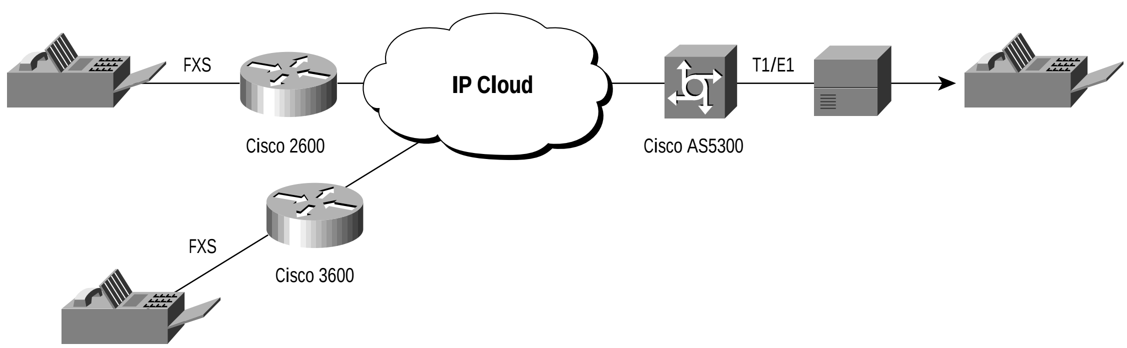
Real-Time Fax to Fax

Companies who have a high volume of fax traffic with partners or suppliers can reduce costs by deploying solutions such as this. In Figure 6, a partner uses a standard fax machine and sends a fax through a Cisco 3600 over a WAN to another fax machine connected to a Cisco AS5300/Voice Gateway through a PBX. A supplier uses a standard fax machine and sends a fax through a Cisco 2600 over a WAN to another fax machine connected to a Cisco AS5300/Voice Gateway through a PBX.

The fax relay feature automatically determines when an incoming call is a fax transmission, signals the receiving fax machine over the IP network, and then accepts the fax transmission for delivery only when the receiving fax machine is on line. The end user can, therefore, be sure that the fax is delivered.

The AS5300/Voice Gateway voice/fax feature card interfaces with the source and target fax machines by appearing to be a fax machine, using T.30 spoofing. Incoming fax transmissions are demodulated before relay so that the 64K pulse-code modulated fax call consumes only 9.6K or 14.4K of bandwidth across the data network per second of fax transmission.

Figure 6   Real-Time Fax to Fax



Cisco Voice Manager

Cisco Voice Manager (CVM) is a Web-based network management application that configures and monitors Cisco voice-over-IP (VoIP) gateways. Network administrators can implement dial plans, monitor call activity and call-quality parameters in real time, and produce detailed reports showing call history and delivered call quality. The AS5300/Voice Gateway provides a full suite of general and voice-specific Simple Network Management Protocol (SNMP) Management Information Base (MIB) variables. Cisco Voice Manager automatically detects voice-supported products, includes tools to troubleshoot network problems, and provides valuable mandatory call-history information. CVM is ideal for managing up to 50 gateways for enterprise or medium network applications.

Figure 7 Cisco Voice Manager

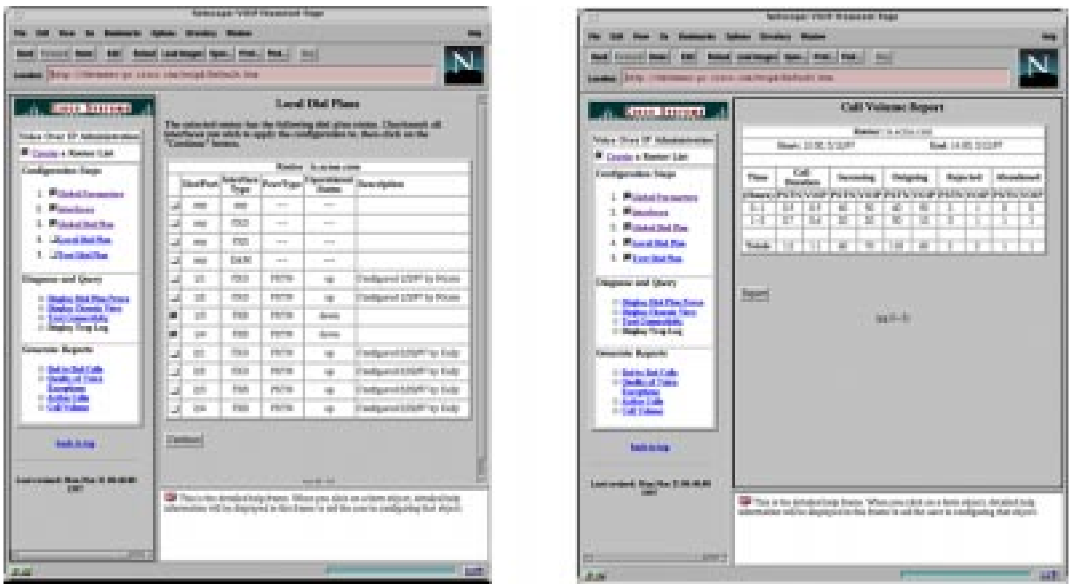


Table 2 Features and Benefits Summary

Feature	Benefit
Digital Interfaces (T1/E1 PRI)	This feature enables higher port densities for increased scalability.
H.323 Compatibility	The Cisco AS5300/Voice Gateway gateways interoperate with H.323-compliant voice and videoconferencing applications such as Microsoft NetMeeting, as well as third-party H.323-compliant gateways and gatekeepers.
High-Performance DSPs	High-performance (100 mips) DSPs have plenty of horsepower to support the full range of available high-compression/low-delay CODECs, including G.711, G.729, G.729a, and G.723.1. The DSPs fully support integrated echo cancellation, voice activity detection, silences suppression, jitter buffering, and comfort noise generation to ensure uniform, high-quality voice conversations. Full coverage for all voice channels is provided.
Low-Latency Design	The Cisco AS5300/Voice Gateway is designed for minimal latency, which is essential for high-quality voice and fax traffic.
Real-Time Fax Relay	This feature enables real-time fax transmission between Group III fax machines operating at up to 14,400 bps. The Cisco AS5300/Voice Gateway voice feature card interfaces with the source and target fax machines by appearing to be a fax machine. Real-Time Fax Relay accommodates delays in the networks using T.30 spoofing. Incoming fax transmissions are demodulated before relay so that the 64K pulse-code modulated fax call consumes only 9.6 Kbps or 14.4 Kbps of bandwidth across the data network per second of fax transmission.
Interoperates with Cisco 3600 and Cisco 2600 Voice Modules	Cisco offers a family of scalable packet voice solutions for businesses. Cisco AS5300/Voice Gateway also interoperates with other gatekeepers.
Multiclass Multilink PPP Fragmentation and Interleaving	This feature reduces large data packets into small packets, prevents voice packets from being stuck behind large packets, and reduces latency.
Cisco Voice Manager	CVM is a configuration, monitoring, and reporting application for packet telephony gateway networks. It provides call history reports, call volume reports, and quality of voice exception reports using the ITU-T G.113 specification for voice quality. This Java application runs on Windows NT or Solaris.

Table 2   Features and Benefits Summary (Continued)

Feature	Benefit
High Performance Co-Processor Design	This is a highly integrated single-device solution that minimizes packet latency, essential for high-quality voice. PC-based solutions using loosely coupled components cannot achieve the same performance characteristics.
Modular Architecture	The system provides flexibility and investment protection. The Cisco AS5300/Voice Gateway modular design allows for scaling from 24 to 120 voice connections per device. In addition, modem and voice modules can both be used in the same AS5300, giving customers added flexibility.
Compatible with Existing Phones, Faxes, PBXs, and Key Systems	This feature provides a standard interface to your existing telephony equipment. Users continue to use familiar equipment, with no special adaptation or retraining.
Real-Time CODEC Selection	This sophisticated DSP architecture supports simultaneous H.323 capability negotiation on all channels. The Cisco AS5300/Voice Gateway voice feature card loads appropriate voice or fax CODEC on the fly for all 120 channels, simultaneously if necessary.
E.165 Echo Cancellation	This feature provides echo cancellation into the circuit-switched network with a tail of up to 32 msec, more than adequate to support carrier quality.
Adaptive Jitter Buffering	Adaptive jitter buffer intelligently balances delay and packet loss through the gateway for maximum call clarity and quality.
Voice Activity Detection (silence suppression)	Bandwidth on the packet network is used only when someone is speaking. During silent periods of a phone call (up to 50 percent of the time), bandwidth is available for data traffic.
Front-End Clipping (Time Before Speech Activity is Detected After a Period of Silence)	0 msec
Hang-Over Time (Maximum Time Before Silence is Recognized After a Period of Speech)	200 msec
Comfort Noise Regeneration	To better simulate phone calls over voice networks, this feature reassures the phone user that the connection is being maintained, even when no voice conversation is in progress.
Voice Quality Statistics	Call parameters used for the ITU-T G.113 recommendation for voice quality impairment calculations are supplied. These include CODEC type, bandwidth used, end-to-end delay, circuit noise, loudness, echo, packet loss, and other statistics.
ITU Standard CODECs G.711, G.729, G.729a, and G.723.1	These standards-based compression technologies, allow for high-quality voice and compression as low as 53kbps to minimize bandwidth required to transmit packet telephony.
Compressed Real-Time Protocol (CRTP) and Multilink PPP Fragmentation and Interleave	These are header compression and packet fragmentation techniques that allow toll-quality voice and fax transmissions over low bandwidth WAN connections.
Dial Plan Mapping	This feature entails mapping of dialed phone numbers to IP addresses. Mapping can be programmed directly in the Cisco AS5300/Voice Gateway or alternately maintained in H.323 gatekeepers that communicate to multiple gateways via H.323 RAS messages.
Number Expansion	This feature enables speed dialing and simplifies dial plan configuration. It expands all numbers matching a defined pattern, so you need to configure and dial only the last few significant digits of the number.
Direct Inward Dial	Direct Inward dial allows direct dialing of each user sitting behind a PBX; there is no need to dial the main number and then dial an extension. This feature is also useful in service provider applications with a VCO/4K used as a service node.
Secondary Dial Tone	This feature allows explicit access to a VoIP network for two-stage calling implementation. The first dial tone is generated by the local phone company and the second dial tone, or prompt, is generated by the VoIP carrier.
Call Progress and Tone Generation	This feature generates call progress tones, including dial, busy, ring-back, and congestion tones, with local country variants.
Dual Tone Multifrequency (DTMF) Transport	This feature enables the use of touch tones for voice-mail applications and IVR systems; it also supports out-of-band DTMF relay when high-compression CODECs such as G.723.1 are used, which may corrupt inband DTMF tones.
Fax Autodetect	Any port can accept a fax relay call; the port is automatically reconfigured when an incoming fax call is detected. A scalable design gives the Cisco AS5300/Voice Gateway the unique ability to reload all fax algorithms simultaneously, meeting the stringent timing requirements of legacy fax machines.
-law and A-law Encoding on Any Channel	This feature facilitates international calling by transparently transcoding between -law encoding (used in T1 countries) and a-law (used in E1 countries).
Music on Hold Threshold	This feature offers intelligent music on hold handling.

Table 3 Cisco IOS Software and QoS Features and Benefits Summary

Feature	Benefit
IP Precedence	The Type of Service (ToS) field in an IP header provides three settable precedence bits. Voice traffic should be configured with IP precedence of 5. This capability allows the network to prioritize voice packets above other traffic, thereby assuring highest voice quality over congested network.
Queuing Mechanisms	Queuing mechanisms include WFQ, priority queuing and custom queuing. These are configurable Cisco IOS software capabilities that reserve appropriate bandwidth and prioritize voice and fax traffic to ensure transparent delivery of toll-quality voice and fax.
CAR	The committed access rate (CAR) feature performs both packet classification and bandwidth management functionality. The packet classification features let users partition network traffic into multiple priority levels or classes of service (CoSs). The network operator can define up to six CoSs using the three precedence bits in the ToS field in the IP header. The operator can then use the other QoS features to assign appropriate traffic-handling policies, including congestion management, bandwidth allocation, and delay bounds for each traffic class.
WRED	WRED features provide network operators with powerful congestion-control capabilities designed to provide preferential treatment for premium-class traffic under congestion situations while concurrently maximizing network throughput and capacity utilization and minimizing packet loss and delay.
RSVP	RSVP allows Cisco voice gateway to request/ reserve required bandwidth for a call.

Table 4 Cisco AS5300/ Voice Gateway Feature Card Technical Specifications

Processor Type	mips 4700 CPU @ 100MHz
Memory	4 MB DRAM
Flash Memory	8 MB
Digital Signal Processor	TMS320VC549 @ (100 mips)

The Cisco AS5300/Voice Gateway includes three expansion slots. One slot is for a Quad T1/E1/PRI feature card and the other two can be used for voice or modem feature cards. Since a single Voice Feature Card can now support up to 48/60 voice calls, the Cisco AS5300/Voice Gateway system can support a total of 96/120 voice calls.

The default memory configuration for Cisco AS5300/Voice Gateway is with 16MB of system Flash and 64MB of shared DRAM. The use of these Voice Feature Cards also requires IOS release 12.0.2XH or later.

Ordering/Configuration Details

For ease in ordering, simply order AS5300-96VoIP-A. It is a pre-configured Cisco AS5300/Voice Gateway (included parts shown in Table 5). AS5300-120VoIP-A is the (E1) 120 voice channel counterpart.

Table 5 AS5300-96VoIP-A includes:

Product Number	Description
AS5300	5300 base unit
SF53-CVP-12.0.2XH	Cisco AS5300 Series IOS IP Voice Plus
AS53-T1-96VOXD	96 Voice channels and Quad T1/PRI bundle
VC-SWA-4.0	All Voice Codec feature set firmware
MEM-16F-AS53	Flash memory upgrade from 8 MB to 16 MB
MEM-64M-AS53	Main DRAM upgrade 32 MB to 64 MB
AS53-AC-PWR	AC power

Note: If DC is required, AS53-DC-PWR can be ordered as an additional option. If redundant power is required, AS53-AC-RPS or AS53-DC-RPS can be ordered as an additional option.

If you already use a Cisco AS5300 for modem dialup and you want to make it voice capable, order:

AS53-CC-48VOXD=	Voice/fax carrier card 48 channels of voice support, voice firmware
FLV-53=	Voice feature license
SF53-CVP-12.0.2XH	Cisco AS5300/Voice Gateway Series IOS IP Voice Plus

If you already use a Cisco AS5300/Voice Gateway for voice and want to add more voice channels, order:

AS53-CC-48VOXD=	(Voice/fax carrier card, 48 channels of voice support, voice firmware).

AS53-CC-60VOXD= is available for E1 systems.

Note: These voice feature cards, based on c549 DSP modules, cannot be mixed with cards based on the prvious c542 based cards. For example, AS53-VOXD DSPMs cannot co-exist in the same chassis with AS53-GVOX DSPMs.

Table 6 Cisco AS5300/Voice Gateway Technical Specifications

Standard Specifications	
Processor Type	150-MHz R4700
Memory	64 MB DRAM
Flash Memory	16 MB system Flash, single or dual bank, up to 16 MB boot Flash
Chassis Slots	Three
Ethernet (RJ-45)	Two (one 10 MB, one 10/100 MB)
Voice/Fax Ports	Up to 96 (T1) or 120(E1)
56K Modems	Up to 48 (T1) or 60 (E1) modems when 48 (T1) or 60 (E1) voice ports are installed
ISDN PRI, T1, or E1	Supports PRI Q.931 and channel-associated signaling
Other Standard Components	Power supply and cord, console cable, two RJ-48C cables, carrier card tool
Dimensions (H x W x D)	3.4 x 17.5 x 18.25 in. (xx cm [need to provide])
Weight	32 lb (19 kg)
Environmental Conditions and Power Requirements	
Operating Temperature	32 to 104 F (0 to 40 C)
Nonoperating Temperature	–40 to 185 F (–40 to 85 C)
Operating Humidity	5 to 95%, noncondensing
Noise Level	34 dB @ 3 ft (0.914 m)
Input Voltage, AC Power Supply	100 to 240 VAC2
Current	2 to 5A
Frequency	50/60 Hz
Input AC Power	200 to 400W (maximum)
Input Voltage, DC Power Supply	–48 to –60 VDC
Maximum Input Current	9.0A
Typical Input Current	3.0 to 4.0A
Input DC Power	200 to 400W (maximum)

Standard Specifications	
Protection	Current limit, overpower, over temperature
Typical Output Power	350W
WAN Interface Options	• Quad T1/PRI (RJ-45); Quad E1/PRI (RJ-45)
Auxillary Interfaces	• Console and Auxiliary Ports • Asynchronous serial (RJ-45)

Table 7 Safety and Regulatory Compliance

Item	Specification
Electromagnetic Compliance (emissions)	EN 55022B
	AS/NZS 3548 B
	VCCI B
	CFR47 Part 15 B (FCC)
Electromagnetic Compliance (immunity)	61000-4-2 (electrostatic discharge)
	61000-4-3 (radiated immunity)
	61000-4-4 (electrical fast transients)
	61000-4-5 (surge)
	61000-4-6 (conducted immunity)
Safety Certifications	UL 1950, third edition
	CSA 950, third edition
	EN 60950, with amendments 1, 2, and 3
	IEC 950
	AS/NZS 3260
	TS 001

Item	Specification
PTT Certifications	CTR4
	CTR 12/13
	JATE
	TS 014
	SS 63 63 34
	PD 7024
	BE/SP-103
	HKT CR11 and CR13
	FCC Part 68
	CS-03
NEBS Compliance	The Cisco AS5300/Voice Gateway is NEBS Level 3 compliant.

Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters
Cisco Systems Europe s.a.r.l.
Parc Evolic, Batiment L1/L2
16 Avenue du Quebec
Villebon, BP 706
91961 Courtaboeuf Cedex
France
http://www-europe.cisco.com
Tel: 33 1 69 18 61 00
Fax: 33 1 69 28 83 26

Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
http://www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

Asia Headquarters
Nihon Cisco Systems K.K.
Fuji Building, 9th Floor
3-2-3 Marunouchi
Chiyoda-ku, Tokyo 100
Japan
http://www.cisco.com
Tel: 81 3 5219 6250
Fax: 81 3 5219 6001

Cisco Systems has more than 200 offices in the following countries. Addresses, phone numbers, and fax numbers are listed on the
Cisco Connection Online Web site at http://www.cisco.com/offices.

Argentina • Australia • Austria • Belgium • Brazil • Canada • Chile • China • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE Finland • France • Germany • Greece • Hong Kong • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Singapore Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey • Ukraine • United Kingdom • United States • Venezuela

6/99 B&W