Data Sheet

Cisco 2600, 3600 and 3700 Voice Gateway Router Router Interoperability with Cisco CallManager

Cisco 2600, 3600 and 3700 multiservice platforms can be deployed as Voice Gateway Routers as part of the Cisco AVVID (Architecture for Voice, Video and Integrated Data)-enabled Cisco CallManager IP telephony solution. New and existing deployments can benefit by using 2600/3600/3700 multiservice platforms as Voice Gateway Routers with Cisco CallManager. Cisco 2600, 3600 and 3700 Voice Gateway Routers communicate directly with Cisco CallManager, allowing for the deployment of IP telephony solutions that are ideal for large enterprises and service providers that offer network managed services. The Cisco 2600/3600/3700 Voice Gateway Router capability leverages an award-winning modular platform that is designed to provide a highly flexible, scalable, multiservice solution for small and medium-sized branches and regional offices. The Cisco 2600/3600/3700 Voice Gateway Router supports the widest range of packet telephony- based voice interfaces and signaling protocols within the industry, providing connectivity support for over 90 percent of the world's private branch exchanges (PBXs) and public switched telephone network (PSTN) connection points. Signaling support includes T1-PRI, E1-PRI, T1-CAS, E1- R2, T1/E1 QSIG, T1 FGD, BRI, FXO, E&M and FXS. The Cisco 2600, 3600 and 3700 Voice Gateway Routers can be configured to support from two to 300 voice channels. As enterprises seek to deploy an expanding list of IP telephony applications and services, Cisco 2600, 3600 and 3700 Voice Gateway Routers, interoperating with Cisco CallManager, provide a solution that will grow with the changing needs of enterprises.

Interoperability Using H.323 or MGCP

The Cisco 2600/3600/3700 Voice Gateway Routers can communicate with the Cisco CallManager using H.323 or Media Gateway Control Protocol (MGCP):

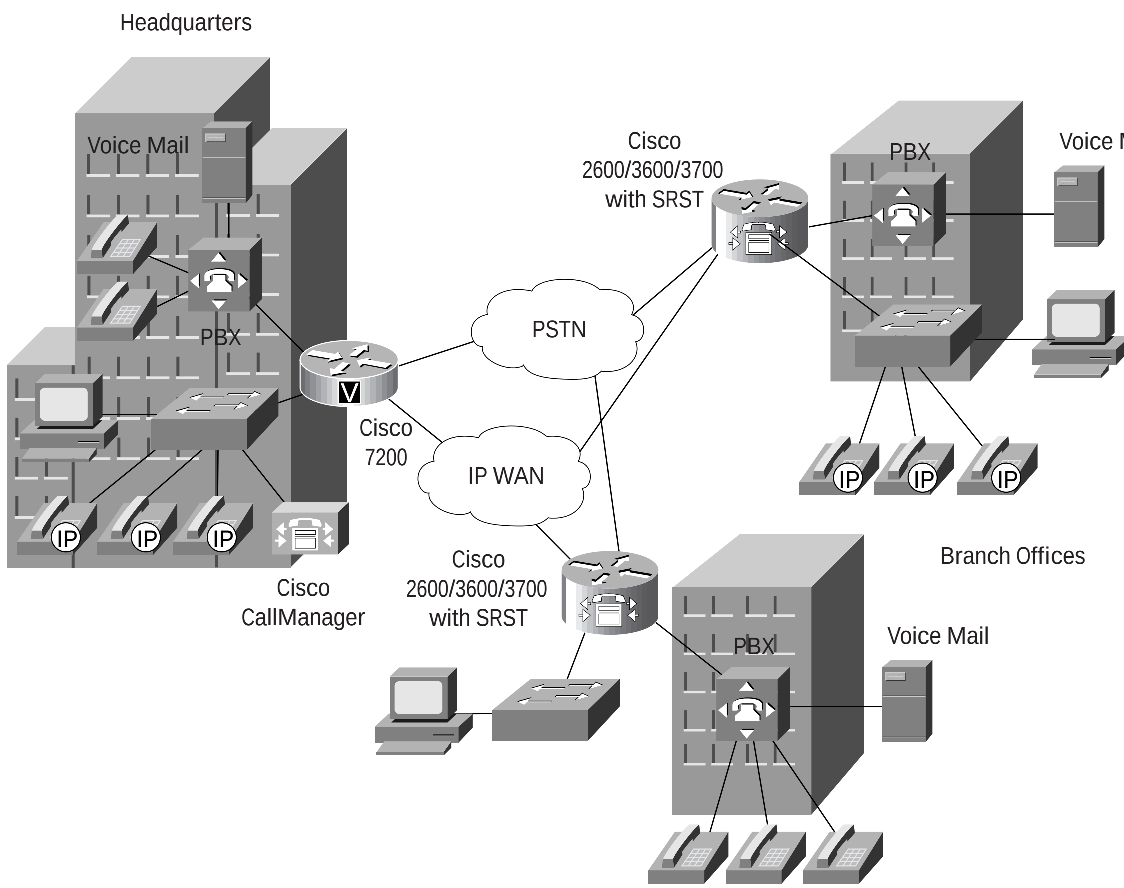
- In H.323 mode, the Cisco 2600/3600/3700 Voice Gateway Router communicates with Cisco CallManager as an intelligent gateway device.
- In MGCP mode, the Cisco 2600/3600/3700 Voice Gateway Router operates as a stateless client, giving Cisco CallManager full control.

As MGCP Voice Gateway Routers, Cisco 2600/3600/3700 multiservice platforms provide enhanced network management and failover capabilities. In MGCP mode, dial plans are configured centrally in Cisco CallManager, instead of in each gateway. And all Cisco 2600/3600/3700 MGCP Voice Gateway Routers in a Cisco AVVID-enabled IP telephony network can be automatically configured by downloading XML files from Cisco CallManager. Also, Cisco 2600/3600/3700 MGCP gateways provide multiple levels of failover capabilities, including Survivable/Standby Remote Site Telephony support to prevent call-processing interruptions or dropped calls in the event of a Cisco CallManager or WAN failure.

IP Telephony Phased Migration

The Cisco 2600/3600/3700 Voice Gateway Router enables users to immediately deploy an end-to-end IP telephony network architecture or gradually shift voice traffic from traditional circuit-switched networks to a single infrastructure carrying data, voice, and video over packet networks. Initially, customers can use the Cisco 2600/3600/3700 Voice Gateway Routers to interconnect legacy PBXs over the packet infrastructure and still maintain PSTN (off-net) connectivity via their circuit-switched PBXs. Later, customers can migrate PSTN (off-net) connectivity to the Voice Gateway Routers and start to incorporate IP phones at larger sites (Figure 1). After all sites are running IP telephony, users can begin deploying IP-based applications such as IP unified messaging, personal assistants, and extension mobility. The Cisco 2600/3600/3700 Voice Gateway Router is an ideal solution for circuit-switched PBX and PSTN access within a Cisco CallManager IP telephony architecture.

Figure 1
IP Telephony Phased Migration—Migrate Circuit-Switched PSTN and PBX Connectivity to Voice Gateway Routers



As companies seek to deploy IP telephony solutions across the entire enterprise—converging voice, video, and data across potentially thousands of sites—they require a feature-rich IP telephony solution that offers simple administration, virtually unlimited scalability and high availability. The Cisco 2600/3600/3700 MGCP Voice Gateway Routers work in concert with the Cisco CallManager, deployed in either a distributed or centralized call-processing model, to provide the IP telephony solutions that enterprises require.

Centralized Call-Processing

Demand for technology to support increased employee productivity and lower costs is at an all-time high. At the same time, many organizations are struggling to deploy new applications and services because of flat budgets. The centralized call-processing model can provide technology to users who require it, while simultaneously providing ease of centralized management and maintenance of applications to network administrators. Instead of deploying and managing key systems or PBXs in small offices, applications are centrally located at a corporate headquarters or data center, and accessed via the IP LAN and WAN. This deployment model allows branch office users to access the full enterprise suite of communications and productivity applications for the first time, while lowering total cost of ownership (TCO). There is no need to "touch" each branch office each time a software upgrade or new application is deployed, which accelerates the speed in which organizations can adopt and deploy new technology solutions. In the Internet economy, the ability to quickly roll out new applications to remote users can provide a sustainable competitive advantage versus companies that must visit each of their branch sites to take advantage of new applications. An architecture in which a Cisco CallManager and other Cisco AVVID applications are located at the central site has the following benefits:

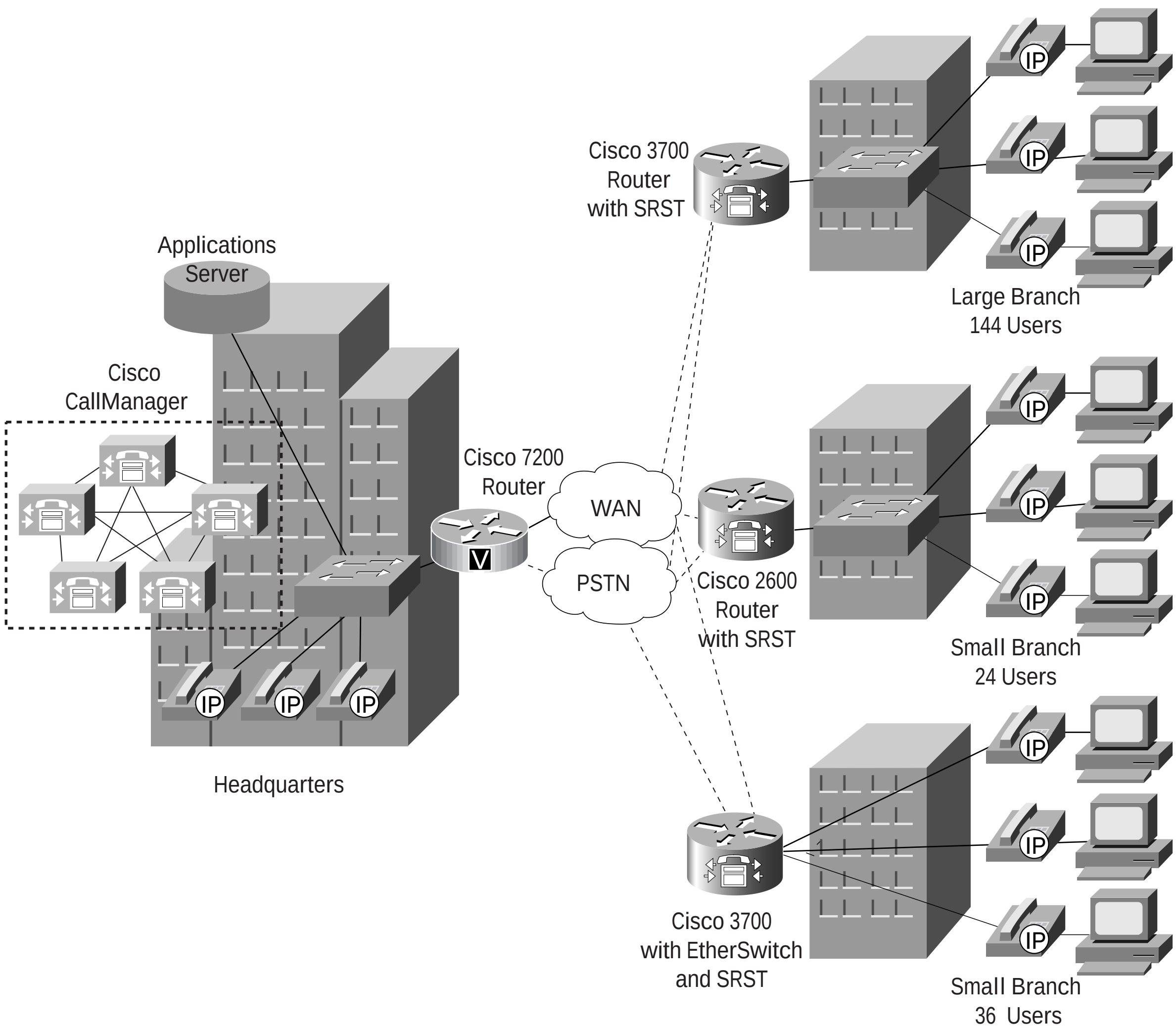
- Centralized configuration and management
- Access at every site to all Cisco CallManager features, next-generation contact centers, unified messaging services, personal productivity tools, mobility solutions, and soft phones all the time
- IT staff is not required at each remote site
- Ability to rapidly deploy applications to remote users
- Easy upgrades and maintenance
- Lower TCO

Survivable/Standby Remote Site Telephony (SRS Telephony)

As enterprises extend their IP telephony deployments from central sites to remote offices, an important consideration is the ability to cost-effectively provide failover capability at remote branch offices. However, the size and number of these small-office sites preclude most enterprises from deploying dedicated call-processing servers, unified messaging servers, or multiple WAN links to each site to achieve the required high availability. The Cisco CallManager IP telephony solution with SRS Telephony allows companies to extend high-availability IP telephony to their remote branch offices with a cost-effective solution that is easy to deploy, administer, and maintain.The SRS Telephony capability is embedded in the Cisco IOS¨ Software that runs on the Cisco 2600/3600/3700 Voice Gateway Router. SRS Telephony software automatically detects a connectivity failure between Cisco CallManager and IP phones at the branch office. Using the Cisco Simple Network Automated Provisioning (SNAP) capability, SRS Telephony initiates a process to intelligently auto-configure the Cisco 2600/3600/3700 Voice Gateway Router to provide call-processing backup redundancy for the IP phones in the affected office. The router provides essential call-processing services for the duration of the failure, ensuring that critical phone capabilities are operational. Upon restoration of the connectivity to the CallManager, the system automatically shifts call-processing functions back to the primary Cisco CallManager cluster. Configuration for this capability is done only once in the Cisco CallManager at the central site.

Figure 2

Centralized Cisco CallManager Deployment with SRS Telephony



Key 2600/3600/3700 MGCP Voice Gateway Router Features and Benefits

Simple Administration

- Provides centralized administration and management
- Enables administration of large dial plans
- Provides a single point of configuration for a Cisco AVVID-enabled IP telephony network

Availability

- Provides Cisco CallManager redundancy
- Provides call preservation for gateway calls when the host Cisco CallManager fails
- Provides MGCP gateway fallback to H.323 control for basic call-handling when the WAN fails

Scalability

- Meets enterprise office requirements of small offices to large corporations
- Scales up to 30,000 users per cluster with Cisco CallManager clustering.

Investment Protection

- Provides a modular platform design with a growing list of more than 90 interface combinations
- Allows users to increase voice capacity while leveraging their existing investment in Cisco 2600, 3600 and 3700 multiservice platforms

Voice Gateway Router with Cisco CallManager Minimum System Requirements

Table 1 Cisco 2600, 3600, 3700, Access Gateway Module and VG200

Signaling	MGCP	H.323
Analog (FXS, FXO)	Cisco IOS: 12.2(4)T Cisco CallManager: 3.0(5a)	Cisco IOS: 12.2(1)M Cisco CallManager: 3.0(5a)
BRI	Cisco IOS: 12.2(15)ZJ Cisco CallManager: 3.3(2)FP1 (This is a controlled release version of Cisco CallManager and is the only release that currently provides this feature.)	Cisco IOS: 12.2(1)M Cisco CallManager: 3.0(5a)
T1 CAS, T1/E1 PRI	Cisco IOS: 12.2(11)T Cisco CallManager: 3.1(1)	Cisco IOS: 12.1(2)T Cisco CallManager: 3.0(5a)
T1/E1 QSIG	Cisco IOS: 12.2(11)T Cisco CallManager: 3.3(2)	Cisco IOS: 12.1(2)T Cisco CallManager: 3.0(5a)

1. The Cisco 2691, 3725 and 3745 is supported with Cisco IOS 12.2(8)T1 or later and Cisco CallManager 3.2(2c)spA or later.
2. The Access Gateway Module is supported with Cisco IOS 12.2(13)T or later and Cisco CallManager 3.2(2c)spB or later.
3. MGCP is not supported for BRI on the VG200 or Access Gateway Module.
4. The actual DRAM and Flash requirements may vary depending on the particular platform and version of Cisco IOS software used. Please refer to the release notes for the version of Cisco IOS software being used for the exact Flash and DRAM requirements.
5. Network module support and required software may vary. Please refer to the Cisco IOS Release Notes and Cisco CallManager Release Notes for definition of features supported.
6. The IP Communications Voice/Fax Network Module (NM-HD) requires Cisco IOS 12.2(15)ZJ. Cisco CallManager 3.3(3) is required for MGCP support.

Voice Gateway Router with Cisco CallManager Feature Summary

Table 2 Voice Gateway Router with Cisco CallManager Feature Summary

MGCP	H.323	Feature	Benefits
X	X	Analog FXS (Foreign Exchange Station) interfaces loop-start and ground-start signaling	Enable direct connection to phones, fax machines, and key systems
	X	Analog E&M (wink, immediate, delay) interfaces	Enable direct connection to a PBX
X	X	Analog FXO (Foreign Exchange Station) interfaces loop-start and ground-start signaling	Enables connecting to a PBX or key system and provides off-premise connections to/from the PSTN/PTT
	X	Analog (Direct Inward Dial) DID	Enable connection to PSTN
	X	Analog CAMA	Enables PSTN connection for 911 Support
	X	BRI Q.931 network side (NET3)	Enables connection to a PBX or key system
X	X	BRI Q.931 user side (NET3)	Enables connection to PSTN
	X	BRI Q.SIG	Enables connection to a PBX or key system
X	X	T1-CAS E&M (wink start and immediate start) interfaces	Enables connection to a PBX or key system
	X	T1-CAS E&M (delay dial) interfaces	Enable connection to a PBX or key system
	X	T1-CAS FGD	Use to connect to a PBX or PSTN
	X	T1-CAS FXO (ground start and loop start) interfaces	Use to connect to PBX or key system and to provide off-premise connections
	X	T1-CAS FXS (ground start and loop start) interfaces	Use to connect to PBX or key system
	X	E1 CAS	Enable connection to a PBX or PSTN
	X	E1 MelCAS	Enable connection to a PBX or PSTN
	X	E1 R2 (more than 30 country variants)	Enable connection to a PBX or PSTN
X	X	T1/E1 ISDN PRI Q.931 interfaces	Use to connect to PBX or key system and to provide off-premise connections to/from the PSTN/PTT
X	X	T1 and E1 Q.SIG	Use to connect to PBX
X	X	Out-of-Band DTMF	Carry DTMF tones and information out of band for clearer transmission and detection.
X	X	Supplementary services	Enable hold, transfer, forward, and conference capabilities from an IP Phone
X		Single point of configuration for a Cisco AVVID-enabled IP telephony network	Centralizes and automates the configuration process for MGCP Voice Gateway Routers by making them configurable on the Cisco CallManager. Configuration information is automatically downloaded at startup and after any configuration change

Table 2 Voice Gateway Router with Cisco CallManager Feature Summary

MGCP	H.323	Feature	Benefits
X		XML configuration files for single-point configuration	MGCP Voice Gateway Router configuration information is stored in XML file format on a Cisco CallManager server and can be easily manipulated with any standard Web browser
X	X	Cisco CallManager failover redundancy	When a MGCP Voice Gateway Router loses contact with the primary CallManager the gateway reregisters with the next available Cisco CallManager of the three on its list
X		MGCP gateway fallback	When the WAN connection to a main site MGCP CallManager is lost, MGCP gateway fallback provides basic call-handling support for PSTN telephony interfaces on a branch office gateway for the duration of the WAN outage
X	X	Multicast music on hold (MOH)	Enables the Voice Gateway Router to deliver music streams from an MOH server to users on on-net and off-net calls
X	X	Tone on hold	Tone indicates when a user is placed on hold
X	X	Caller ID support	Enables the Voice Gateway Router to send the ANI of a caller for display: In MGCP mode, to/from IP phones, FXS and T1/E1 PRI (Caller ID currently not supported on FXO and T1-CAS) In H.323 mode, between IP phones, FXS, BRI and T1/E1 PRI; and from FXO to IP Phones, FXS, BRI and T1/E1 PRI
X	X	Platform voice scalability	Scale from two to 300 voice channels in a single multiservice router solution
X	X	Modular design	Single box supports data and telephony services
X	X	Standards-based PCM encoding	ITU-T G.711 PCM encoding provides 64-kbps analog-to-digital conversion using u-law or A-law.

Table 3 Voice Gateway Router Feature Summary

MGCP	H.323	Feature	Benefits
X	X	Standards-based compression algorithm support	Users can choose to either transmit voice across their networks as uncompressed PCM or compressed from 5.3 kbps to 32 kbps using standards-based compression algorithms (G.729, G.729a/b, G.723.1, G.726, and G.728).
X	X	Fax support	Transmit Group III fax over any voice channel without sacrificing voice-processing resources regardless of compression type being used.
X	X	Voice over IP (VoIP)	Transmit data, voice, and video across a single WAN connection (frame relay, ATM, ISDN, HDLC, or multilink point-to-point protocol [MLPPP]).
	X	Private Line Automatic Ringdown (PLAR)	Provides a dedicated connection to another phone or an auto attendant
	X	Connection trunk	Create permanent connection across the network, often to carry proprietary PBX signaling
	X	Off-premise extension (OPX)	Extend the capability of legacy PBX to off-premise phones. (This applies to toll-bypass applications only)
X	X	Voice Activity Detection (VAD)	Conserve bandwidth during a call when there is no active voice traffic to send
	X	Busy Out	Busy out desired trunk line to PBX when direct WAN or LAN connection to router is down
X	X	Comfort noise generation	While using VAD, the DSP at destination end emulates background noise from on the source side preventing the perception that a call is disconnected
	X	H.323 version 1, 2, 3, 4 support	Use industry-standard signaling protocols for call setup between gateways, gatekeepers, and H.323 endpoints
X	X	Authentication, Authorization, and Accounting (AAA)	Support debit card and credit card (prepaid and postpaid calling card) applications
	X	Interactive Voice Response (IVR) support	Enables automated attendant support, voice-mail support or call routing based on service desired
	X	Automated Attendant (AA)	Uses IVR to provide automated call answering and forwarding services

Telephony Interface Signaling Support

Table 4 Voice Performance on the Cisco VG200 and Cisco 2600, 3600 and 3700 Series Gateways for 12.2.8T

	2610	2610-XM [5]	VG200	2620	2620-XM[5]	2650 [5]	2650-XM [5]	3620	3640	2691 [3]	3725 [4]	3660	3745 [4]
VoIP Performance: Maximum													
Standalone Voice Gateway Router	48[1]	60[1]	60[2]	60[1]	70[1]	90[2]	90[2]	60[2]	96[1]	90[2]	120[2]	300[1]	240[2]
WAN Edge Gateway	24[1]	30[1]	N/A	30[1]	35[1]	60[1]	60[1]	30[1]	50[1]	90[2]	120[2]	200[1]	240[2]
WAN Edge Gateway with cRTP	12[1]	22[1]	N/A	22[1]	25[1]	42[1]	42[1]	24[1]	30[1]	90[2]	120[2]	160[1]	240[2]
VoIP Performance: Maximum Calls Per Second[1]													
	0.5	0.5	1	1	1	2	2	1	2	4	15	4	20
Maximum Physical DS0 Conn													
FXS	12	12	4	12	12	12	12	12	36	12	24	72	48
FXO	8	8	4	8	8	8	8	8	24	8	16	48	32
E&M	4	4	4	4	4	4	4	4	12	4	8	24	16
Analog DID	4	4	4	4	4	4	4	4	12	4	8	24	16
BRI	4	4	4	4	4	4	4	4	12	4	8	24	16
T1/E1 Ports	3	3	2	3	3	3	3	2	6	2/3	4/6	12	8/10
T1 Channels	72	72	48	72	72	72	72	48	144	48/72	96/120	288	192/240
E1 Channels	90	90	60	90	90	90	90	60	180	60/90	120/180	360	240/300

All results represent G.729 switched calls with VAD turned off.Standalone Voice Gateway Router: FE egress, no QoS features, Voice traffic only WAN Edge Gateway: T1/E1 Serial egress, QoS features (LLQ, LFI, TS), Voice (~50% of bandwidth) + Data traffic (~25% of bandwidth)

WAN Edge Gateway with cRTP: T1/E1 Serial egress, QoS features (LLQ, LFI, TS), cRTP, Voice (~50% of bandwidth) + Data traffic (~25% of bandwidth)

Notes:

1. 5% platform CPU utilization reached for this number of voice channels.
2. Physical DS0 connectivity limit reached for this configuration.
3. In 12.2.8T the 2691 supports a maximum 2 T1/E1s DS0 connectivity. In a later release, when the AIM-VOICE-30 is supported, the 2691 can support a maximum of 3 T1/E1 ports.
4. In 12.2.8T the 3725 and 3745 support a maximum 4 and 8 T1/E1s DS0 connectivity, respectively. In a later release, when the AIM-VOICE-30 is supported, the 3725 and 3745 will physically support a maximum of 6 and 10 T1/E1 ports, respectively. The capacity figures at that time may be revised to be higher.
5. 128M of memory equipped

For more information on Cisco 2600/3600/3700 multiservice platforms and Cisco IP telephony, visit: Digital T1/E1 Packet Voice Trunk Network Module Data Sheet: http://www.cisco.com/warp/public/cc/pd/rt/2600/prodlit/st1e1_ds.htm

- Low Density Voice/Fax Network Modules for the Cisco 2600, 3600 and 3700 Data Sheet: http://www.cisco.com/warp/public/cc/pd/rt/3600/prodlit/c36p_ds.htm
- Cisco SRS Telephony: http://www.cisco.com/warp/public/cc/pd/unco/srstl/index.shtml
- Cisco CallManager Version 3.1: http://www.cisco.com/warp/public/cc/pd/nemnsw/callmn/prodlit/callm_ds.htm
- Cisco AVVID for the Enterprise: http://www.cisco.com/warp/public/779/largeent/avvid/cisco_avvid.html

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

European Headquarters
Cisco Systems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

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

Asia Pacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 6317 7777
Fax: +65 6317 7799

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