Multi-Service Access Application (VoIP Mode)

This chapter provides a general description of an application in which a VISM equipped Cisco MGX 8850 equipped provides multi-service access between a customer's TDM network and a Voice Gateway over a packet network.This is an application of VISM operating in VoIP mode.
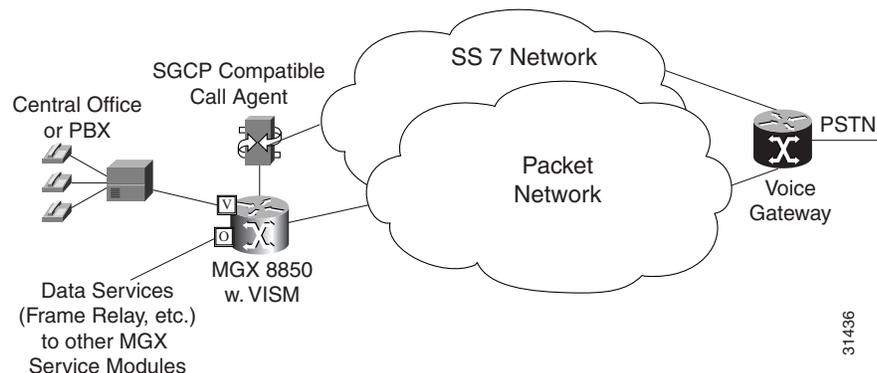
Application Description

Figure 5-1 shows a high level view of voice traffic being conveyed over a packet network using VISM equipped MGX 8850s at each access point to the network.

This application requires that VISM be in the VoIP operating mode.

VISM operates in conjunction with the SGCP compatible call agent and two networks are used. The Call Agent connects to the telephone SS7 network and handles the call control signaling. The VISM connects to an IP network and handles the voice payload between the customer's TDM voice/data network and a Voice Gateway. For transmitting the voice payload to the IP network the application uses Voice over IP (VoIP) segmented in AAL 5 ATM cells. Other data services (such as Frame Relay) can also be accommodated by configuring the MGX 8850 with the appropriate service modules (e.g. FRSM) and using separate PVCs into the packet network.

Figure 5-1   VISM Multi-Service Access Application

When a call is initiated, the signaling is routed from the Central Office or the customer's PBX to the Call Agent. The call is setup under the control of the Call Agent and in conjunction with its local MGX 8850/VISM, the remote Call Agent, and the remote MGX 8850/VISM. When the call setup procedure

is complete each VISM knows the IP address of the other VISM associated with the call and an end-to-end IP bearer circuit is established between the calling and called parties through the Voice Gateway. At this point, the voice conversation can proceed.

Figure 5-2 shows the Multi-service Access arrangement in more detail. The MGX 8850/VISM is connected to the network by an OC-3 line which is used for the voice payload, communication with the VSC 3000, and other data services if equipped. A network edge router routes the voice traffic across the network to the voice gateway handling the called party and routes call control information between the VISM and the Call Agent.

For reliability, two PVCs to the network can be established using separate edge routers. If the primary circuit fails transmission automatically switches to the secondary circuit.

Using all the available slots, the MGX 8850 can be configured with up to 24 VISM cards with the following restrictions.

The two lower shelf cellbuses can each sustain a bandwidth of one OC-3/STM-1 link. For example, this bandwidth limits the number of E1 ports on the lower shelf, when using the G.711 codec, to 78 (approximately 10 VISMs).

Figure 5-2   Voice Gateway using VISM

The TDM Interface

Each VISM supports up to eight T1 lines connected either to a telephone central office or a customer's PBX. Each T1 line conforms to the ITU-G.711 standard and can be configured for B8ZS or AMI coding. Both SF and ESF frame formats are supported with the default being ESF. Line failures cause the VISM to generate T1 facility alarms (AIS,RAI) and lines can be tested using continuity tests and line loopbacks. CAS signaling in the TDM data streams are detected and backhauled to the Call Agent using Simple Gateway Control Protocol.

Voice Signal Processing

Voice coding is Mu law PCM with Voice Activity Detection (VAD) for silence suppression over the network and comfort noise generation at the remote end. By default each DS0 has echo cancellation enabled with the tail set to 128 msec (the tail can be configured from 24 to 128 msec).

Modem and Fax detection are supported. Upon detection of a 2100 Hz tone reversal, echo cancellation and VAD are disabled. While in data mode, the line is monitored for carrier and if carrier is lost for more than 100 msec, the line is restored to its previous codec.

Packet Network Interface

VISM communicates with the packet network for transmitting the voice payload by using the SONET OC-3 port on the MGX 8850 PXM card. Voice payload samples are formatted and sent across the MGX 8850's cell bus and onto the SONET connection.

The payload is first formatted into a RTP packet, encapsulated in an UDP, encapsulated in an IP packet and then converted to AAL 5 ATM cells for transmission to an edge router on the network. RTP is used to allow time-stamping of the voice samples which permits dejittering of samples transmitted to the destination TDM line.

A single permanent virtual circuit is set up between the MGX 8850 and the router and all packets are sent across the PVC regardless of their destination. It is the responsibility of the router to extract the IP address and route the cells accordingly.

To improve reliability, VISM 2.0 supports two independent OC-3 interfaces each connected to a separate edge router and each with its own PVC. One PVC is designated the primary PVC and the other the secondary PVC. The primary circuit is used unless it fails in which case the SONET Automatic Switchover feature switches to the secondary circuit. Switchover may cause a temporary 250 msec glitch on the lines.

If the MGX 8850 is configured with service modules to support Frame Relay and other data services, the OC-3 interface is used for these services also using their own adaption types and PVCs.

Call Agent Interface

VISM has IP connectivity with the call agent which is achieved through the OC-3 connection to the edge router and then onto the call agent.

Special Gateway Controller Protocols are used which are transmitted as UDP/IP datagrams using AAL 5 ATM to the edge router. Part of the initial setup procedure is to assign IP addresses for the VISM and the Call Agent and configure each with the other's IP address as well as their own.

In this application VISM supports:

Simple Gateway Controller Protocol (SGCP) version 1.1
The protocols allow the following functions to be performed:

Call control for the setup and teardown of calls
Resource coordination for auditing the status of VISM and lines

The protocols allow for the backhauling of CAS signaling information from VISM to the Call Agent.

For more details of VISM/Call Agent communications, please refer to the VoIP section in Chapter 3.

Configuration using the CLI

VISM configuration is performed by entering CLI commands either from an alpha-numeric terminal connected to the PXM's serial port or through CiscoView.

Besides the configuration of the VISM card itself, the Call Agent must be configured and a connection must be established between VISM, the MGX 8850 PXM and the Router at the edge of the IP network must be configured.

The following sections describe the commands that are executed to setup the Tandem Switch application. A detailed description of all the VISM commands can be found in Chapter 7.

VISM Card Configuration

Logon to the MGX 8850 shelf using a valid user name and password and enter the slot number of the VISM card to be configured.

> **Note**   If the VISM slot number is not known, enter a slot number of 7 or 8 (this will switch to a PXM card) and execute the display cards command, **dspcds**. The resulting display indicates the slot locations of all the cards in the shelf. Then use the change card command, **cc**, to switch to the VISM card.

Step 1   Basic Card Configuration

    a.   A new VISM card (one that has not been configured) comes up by default in VoIP mode. Check the current mode setting using the **dspvismparam** command. If necessary, set the VISM card to VoIP mode using the **cnfvismmode** command.

    b.   Allocate resources on the card by executing the **addport** command followed by the **addrscprtn** command. Neither of these commands require any parameter values to be entered.

    c.   Configure the domain name of the VISM card using the **cnfvismdn** command. The domain name (maximum length of 64 characters) should be the same as the domain name specified for the VISM card when configuring the Call Agent.

    d.   Configure the IP address of the VISM card using the **cnfvismip** command. When specifying the VISM IP, the netmask must be assigned the value of 255.255.248.0.

    e.   Specify DSP Template
VISM front card contains 37 digital signal processors (DSPs) of which 24 are dedicated to voice compression only, 12 are general purpose (voice compression or echo cancellation) and one is used as a jukebox. As a guide, one DSP has the capacity to handle 21 DS0's with echo cancellation but with no compression. Use the **cnfcodectempl** to specify which DSP template to be used. Template 1 includes G.711, G.726-32, G.729a, G.729ab and clear channel. Template 2 includes G.711 and clear channel.

    f.   Bring the VISM into service using the **cnfgwis** command.

Step 2   Configure VISM for Call Agent Protocol.

Configure the MGCP link to the Call Agent using the following commands:

a. **adddn** - This command specifies a Call Agent domain name and IP address. Using this command repetitively, up to 8 Call Agents can be specified with up to 8 IP addresses for each Call Agent. This command allows the user to specify a preference level when more than one IP address is assigned to the domain name.

b. **addmgc** - This command adds a call agent (by number and domain name) to a table of allowed call agents. This command also allows the user to specify the Call Agent as "Associated" in which case RSIP (restart in progress) messages are sent to the Call Agent.

c. **addmgcprotocol** - This command specifies the gateway control protocol to be used for a specific Call Agent. To configure the SRCP link to the Call Agent select the protocol number for SRCP when executing the **addmgcprotocol** command.

d. The following optional commands can be executed to further specify gateway control protocol parameters.
    cnfxgcpretry for the number of retries
    cnfxgcpmwd for the maximum wait delay
    cnfxgcppeer for MGCP UDP peer port number if other than the default of 2427 is to be used.
    cnfsrcppeer for SRCP UDP peer number if other than the default of 2428 is to be used.

As the Call Agent commands are being entered, VISM populates tables which tie together the Call Agents' names, addresses, MGCP usage and UDP ports as follows. The tables each have 8 rows, one for each Call Agent that have been configured.

Domain Name	IP Addresses 1 to 8								SRCP Y/N?	MGCP Port	SRCP Port
										2427	2478

Step 3   Configure CAC

a. If necessary disable CAC using the **cnfcac** command, the default is enabled.

Step 4   Configure DSPs

a. If the DSP default values are not to be used, configure the echo canceller DSPs parameters using the appropriate **cnfecanxxx** commands. These commands permit the user to specify idle pattern, noise match re-enable, echo canceller re-enable, tone disable, and maximum tail.

Confirm the DSP parameters for each line using the **dsplndsp** command for each line in turn.

Step 5   Bring up T1/E1 lines.

a. Use the **dsplns** command to see which lines are already enabled.To enable additional T1/E1 lines, use the **addln** command for each line entering the appropriate line number (line number corresponds to the port number on the back card).

    b. For each enabled T1/E1 line, use the **cnfln** command to specify the desired coding, length, clock source parameters.

    c. Configure signaling (CAS, CCS, or none) for each line using the **cnfsig** command. Note that only CAS or none are valid for VoIP mode.

Step 6   Specify Endpoints

    a. Use the **addendpts** command to add an endpoint for each T1 or E1 DS0 port. An endpoint is a logical port corresponding to a particular DS0 it is used to specify a connection end and is required by the VSC for sending the SGCP Create Connection and Modify Connection commands to the VISM. The **addendpts** command assigns an endpoint number to each endpoint and links it to a specific line (DS1) and DS0 on that line. The single **addendpts** command can be used to assign endpoints to a group of DS0s.

Configure a VISM/PXM/Router Connection

An ATM AAL5 PVC must be established between the VISM card, the PXM card, and the IP network router.

Step 1   Add and enable a Broadband Interface between the MGX 8850 and the network router.

    a. On the PXM card, use the **addln** and **upif** commands to add and enable the broadband interface. For example, addln -sonet 7.1 and upif 1 1 100 0 4095 (see the MGX 8850 Command Reference manual for details of these commands).

    b. Add the slave segment of the PVC between the VISM and the PXM.

    c. Use the **addcon** command to setup the slave segment. The VPI and VCI values must match those used when setting up the master segment of the same connection in the VISM.

Step 2   Add a PVC from the VISM to the PXM card in the MGX8850

    a. This step involves adding a master PVC between VISM and PXM. Before this is executed, the equivalent slave PVC from the PXM to the VISM must be added.

    b. Use the **addcon** command to create a master segment PVC between the VISM card and the router. Adding a PVC must be done with the CLI and cannot be performed under CiscoView.

    The addcon command has the format:

    addcon [<localVCI>]<preference><pvcType><application><PCR><mastership>[<remoteConnId> where remoteConnId has the format, "nodename.slot.port.vpi.vci".

    When this master segment is setup, the VPI and VCI values specified for the master segment on the VISM card and the slave segment on the PXM card must be the same. Additional PVCs can be added to provide separate PVCs for control and bearer or to provide a second (redundant) channel. If a redundant pair are to be used, use the **cnfconprotect** command to configure the primary and secondary channels.

Step 3   Router Configuration

The router is the point at which the voice bearer channel interfaces with the ATM/IP network. It is the responsibility of the router to route the call through the network to the correct IP destination. The router should be configured so that it recognizes the cells over the ATM link as AAL5 format with encapsulated IP datagrams.

For example, if a Cisco 7500 router is used, the command sequence (after entering configuration mode) would be:

— interface ATM1/0/0.1 point-to-point

— ip address 198.45.16.101 255.255.255.252
— atm pvc ID <Vpi> <Vci> aal5mux ip

— end

— wr (save the configuration) to disk

See the relevant router documentation for the router being used in this application.

Call Agent Configuration

The Call Agent needs to be configured for the gateway control protocol, the bearer and D channel configurations, and the identity of the VISM card. The configuration procedure depends very much on the make and model of Call Agent being used.

Refer to the documentation for the Call Agent being used.