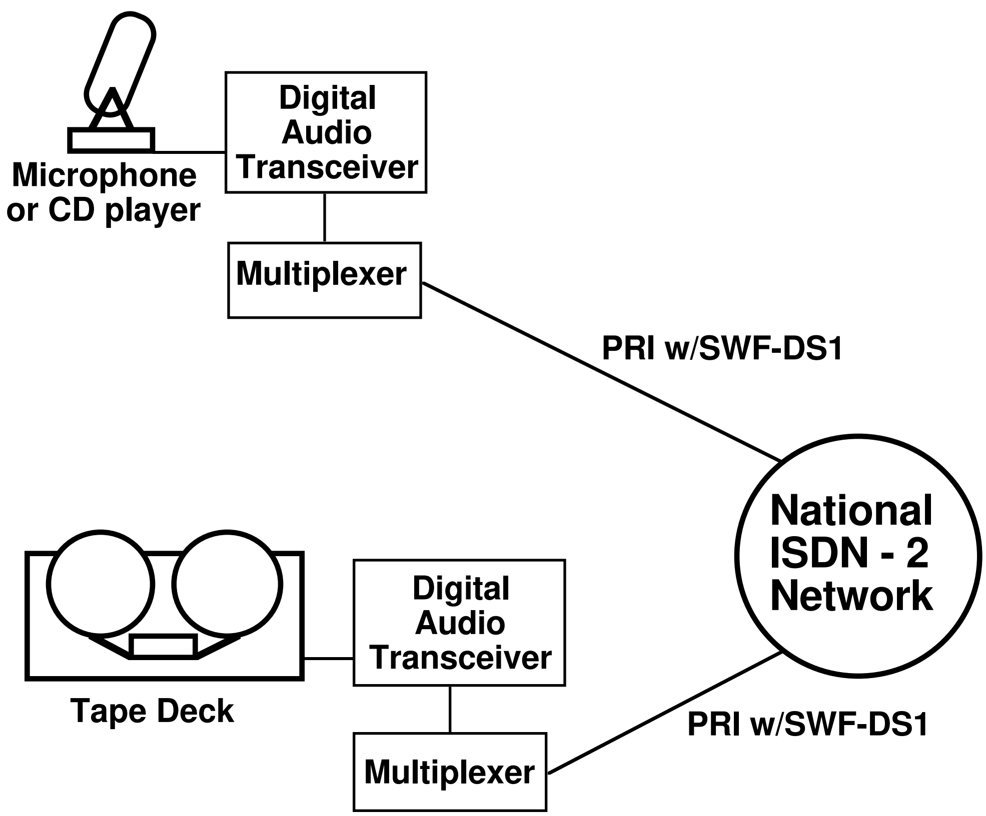
3.3.14 High quality audio transmission

The ISDN network can carry high quality audio signals. Many professional broadcasting applications need the capability of a high quality audio link for applications such as remotely located radio broadcasts. Certain products compress 16-bit audio by a factor of four, essentially using 4-bits for each audio sample. By using ISDN, only four 64 kbps lines are needed in parallel to pass the stereo audio. This can be accomplished using SWF-DS1 (Multirate ISDN) or with BRI and Inverse Multiplexers. Any idle channels can be used (via the multiplexer) for other purposes. Depending on user requirements, quality can range from 7.5 kHz mono to 22.5 kHz stereo by choosing from one to six B channels.

Configuration:



How It Works:

Users place wide bandwidth circuit-data calls between multiplexers using SWF-DS1. Users can select the desired bandwidth of the call on a per-call basis based on the needed quality of the audio transmission. Digital audio transceivers encode and decode the digital signals. Note that for this technology, the audio is always digital through the network; it does not need to be converted to analog prior to transmission. The digitized speech is carried across the all-digital ISDN network without corruption.

Obtain Equipment with These Capabilities

Digital Audio Transceiver

- Audio compression for full duplex audio.
- Interface to a multiplexer as a link to reach the National ISDN network.
- Interfaces to required peripherals - tape decks, microphones, CD players, etc.

Multiplexer

- Interface compatible with digital audio transceiver.
- Supports SWF-DS1 (Multirate ISDN).
- Optional support for framing. Standards are H.221.

Request These Capabilities From ISDN Service Provider

Network Interface to Multiplexer

- One 23B+D PRI.
- Directory Numbers (DNs) as required by application.
- B channels with:

  Circuit data.

  SWF-DS1 (Multirate ISDN).
- **Features for Circuit-Data:**
  - None required.

Technical Notes

- Inverse Multiplexers with multiple BRI lines can also be used to achieve similar functionality for this application.
- Certain products compress 16-bit audio by a factor of four, essentially using 4-bits for each audio sample. Therefore for 15 kHz stereo audio (FM quality), the bit rate is 32,000 (sampling rate) x 2 (stereo) x 4 (bits/sample) = 256 kbps. By using ISDN, four 64 kbps lines are needed in parallel to pass the stereo audio.
- Using the compression assumptions outlined above, the following modes of operation are available for high quality audio:

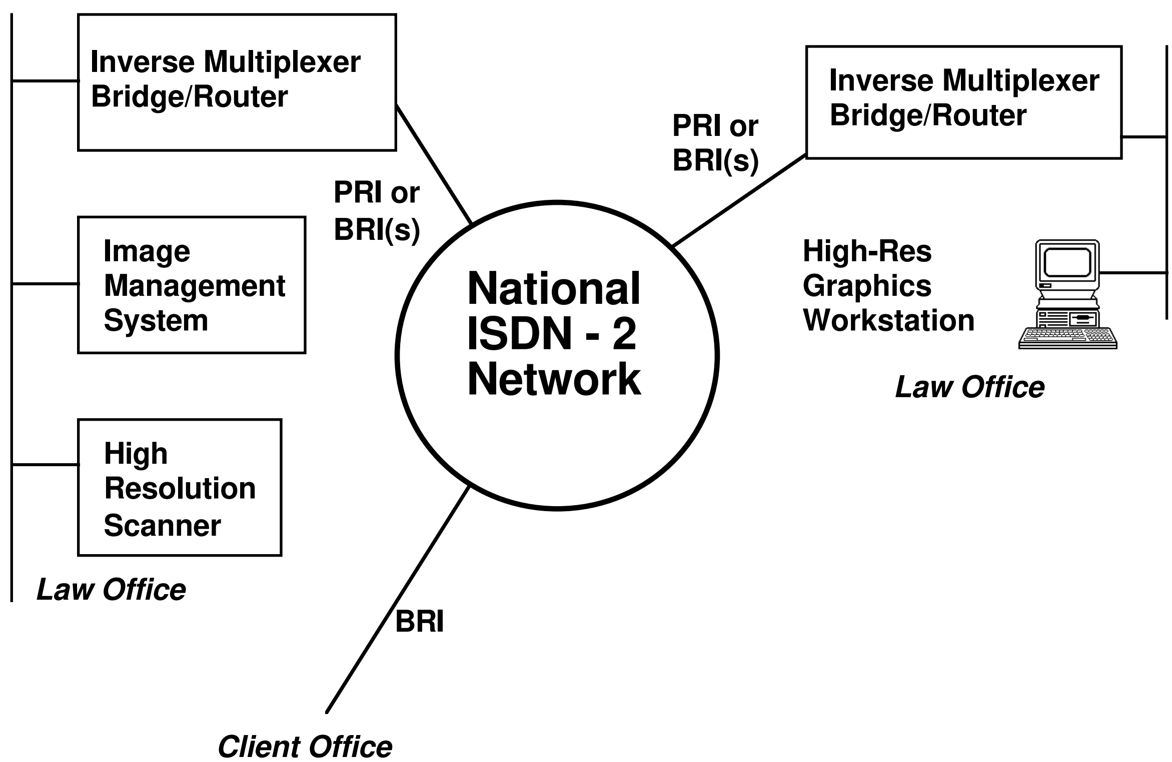
Number of Channels	**Bandwidth**	**Audio quality**
1	64 kbps	7.5 kHz mono
2	128 kbps	7.5 kHz stereo 15 kHz mono
3	192 kbps	11.3 kHz stereo 22.5 kHz mono
4	256 kbps	15 kHz stereo
6	384 kbps	22.5 kHz stereo

- For detailed PRI Signaling Flow Diagrams for this application, see Sections 5.2.3.7 (Multirate Circuit-Data Call Setup) and 5.2.3.8 (Voice, Circuit-Data, and Multirate Circuit-Data Call Clearing).

3.3.15 Access to litigation support system

The combination of LAN and ISDN technology can be used to support attorneys and their staff that may be located off-site at client locations or at remote trial locations with rapid transmission of digitized documentation. Typical document image size 70 Kilobytes, with a required image response time on the order of 1 second. This capability accelerates the research process and eliminates the need to ship documentation.

Configuration:



How It Works:

At each remote location, a Bridge or Router (with an Inverse Multiplexer) or small BRI LAN Bridge or Router can initiate or respond to access from another location. Remote LAN packets are continuously monitored and packets identified for a remote location are transmitted via the Inverse Multiplexer. On receipt of a remote packet, the Inverse Multiplexer immediately establishes one or more circuit switched data connections to the remote location. The appropriate DN(s) are determined by the LAN routing and addressing information. Based on the traffic volume, the Inverse Multiplexer establishes additional circuit switched data connections on one inverse multiplexed logical channel. Alternatively, the network can provide a single wideband channel through SWF-DS1 (Multirate ISDN) on a PRI. As traffic levels decrease the Inverse Multiplexer will release circuit switched data connections.

Obtain Equipment with These Capabilities

Bridge/Router Inverse Multiplexer

- Compatible interfaces with Local LANs (e.g., Ethernet, Token Ring)
- LAN bridging and/or routing functions compatible with Network Operating System and application software.
- Automatic dialing of peer Bridge/Router based on destination address from LAN protocol
- Supports connection to dedicated facility via V.35 to external CSU/DSU, channel bank, T1-MUX , etc.
- Distributes remote traffic across multiple circuit switched data connections and the dedicated facility (e.g., n x 56/64 kbps + 224/256 kbps , respectively) and reassembles data streams at the destination. The multiple connections must support different data rates.
- Inverse multiplexing function to form N x 64k (56k) channel from N individual 64k (56k) circuit-data calls. Calls can be over PRI or multiple BRIs. Alternatively, place SWF-DS1 (Multirate ISDN) calls on a PRI at speeds greater than 64 kbps.
- The Inverse Multiplexer or PRI TA can connect to the network through a PBX or other private network equipment. (These devices support NT-1 and NT-2 functions.)
- Support one or more directory number values for use with circuit-data calls
- Support Calling Number Identification Services for circuit-data calls

Request These Capabilities From ISDN Service Provider

Network Interface to Bridge/Router Inverse Multiplexer (PRI)

- One 23B+D PRI.
- Directory Numbers (DNs) as required by application.
- B channels with:

    Circuit-data.

    SWF-DS1 (Multirate ISDN)

- Features for Circuit-Data:
    - Calling Number Identification Services.

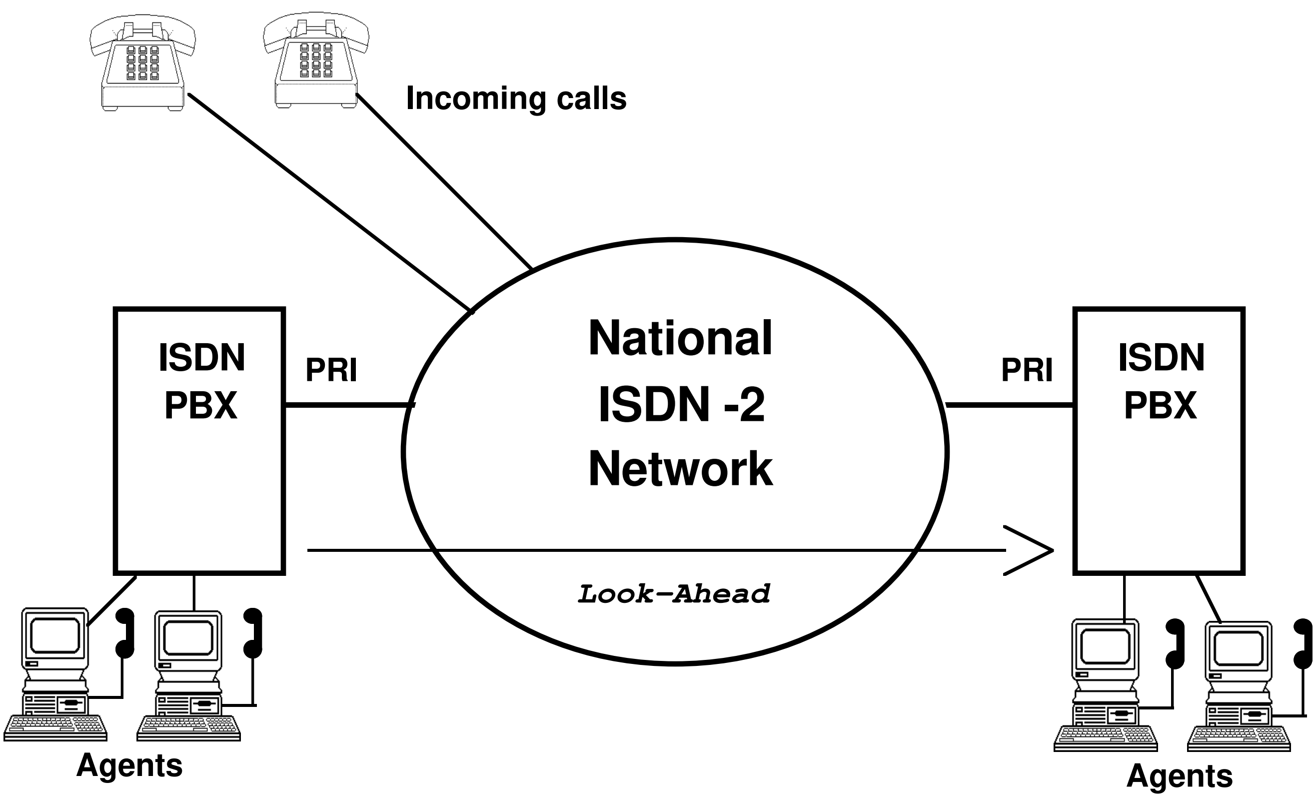
Technical Notes

- The Inverse Multiplexer can utilize a PRI or multiple BRI line network access. The choice of interface will depend chiefly on the total required bandwidth and economics of the services and related CPE (NT-1's, CSU, etc.) -- the services are functionally identical in this application.
- By using PRI with SWF-DS1 (Multirate ISDN), the router can (through a PBX, multiplexer or communications controller) use a single nx64 channel provided by the network. More information on high bandwidth configurations may be found in Section 5.1.9.
- ISDN Bridge/Router Inverse Multiplexers need to use compatible protocols for framing MAC layer packets (e.g., ethernet packets) onto ISDN circuit-data, and compatible protocols for inverse multiplexing.
- Connections to multiple remote locations (LANs) are supported simultaneously with association of LAN addressing and routing information to multiple DN's.
- Calling Number Identification provides additional security to the network.
- For detailed signaling flow diagrams, see Sections 5.2.3.4 (Circuit-Data Call Setup), 5.2.3.7 (Multirate Call Setup), and 5.2.3.8 (Voice and Circuit-Data Call Clearing).

3.3.16 Call center load balancing

A business has two customer service centers to handle incoming calls at different PBXs. To reduce holding times and better manage its services, the company wants to turn these separate facilities into a single virtual call center. Traditionally, interflow of traffic between centers is accomplished by a blind transfer. Using ISDN, the PBX s can determine whether other locations can handle extra calls before forwarding the call. The remote location has the ability to accept or deny the call before it is transferred. Intelligent routing with ISDN improves efficiency, saves valuable agent work time, and improves customer services.

Configuration:



How It Works:

When a preset threshold of queued calls is reached and a new call arrives, that location checks to see if another location can handle the call. While checking, the call stays queued at the PBX where it arrived. The PBXs use ISDN signaling to find out if a agent is free elsewhere. If so, the call is forwarded, together with the original number that the caller dialed so the agent that eventually handles the call can provide the appropriate service. The means of communication between systems is the User-to-User Information (UUI) in ISDN signaling.

Obtain Equipment with These Capabilities

Private Branch Exchange (PBX)

- Automatic Call Distributor (ACD) features, including:
  - Look-ahead interflow
  - Forwarding the original called number.
- Primary Rate Interface, with voice calling on each B channel.

Agent terminals

- Access to information needed to handle forwarded calls, including display/processing of the original called number.

Request These Capabilities From ISDN Service Provider

Network Interfaces to the PBX

- One 23B+D PRI.
- Directory Numbers (DNs) as required by application.
- B channels with:

  Voice
- Features for voice:
  - Calling Number Identification Services
  - Forwarding the called number.
  - User to User Information (UUI) transfer, a proprietary feature. (See Technical Note)

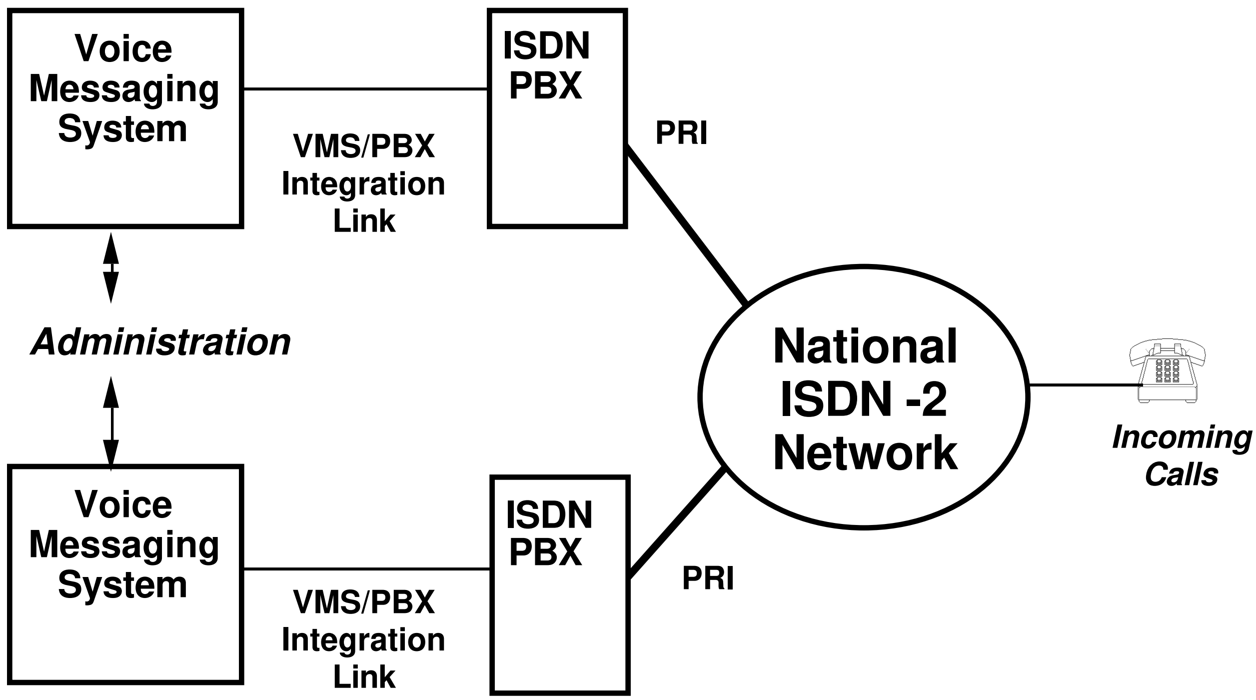
Technical Notes

- User to user information transfer is a feature of each switching system's PRI capability. The feature is offered by service providers. However, NI-2 does not assure interoperable user-to-user information transfer. Each PBX must have PRI access onto switching systems with compatible UUI treatment. Similarly, PBXs that implement the application do so in a proprietary manner.
- For detailed signaling flow diagrams, see Section 5.2.3.3 (Voice Call Setup) and 5.2.3.8 (Voice and Circuit-Data Call Clearing).

3.3.17 Networking Voice Messaging Systems

Networking Voice Messaging Systems with ISDN allows multiple, geographically-separated systems within a centrally administered environment to appear as a single system to the client user. Within the networked Voice Messaging System, the client user will know the caller's identification from the message envelope information and be able to respond with voice messaging sent directly to the caller's mail box. Messages from callers outside of the networked Voice Messaging System will appear as "outside caller" messages and there will be no capability for direct voice messaging response.

Configuration:



How It Works:

When a call is made to a VMS client user attached to the PBX, and the line is busy or there is no answer, the call will be forwarded to the VMS serving the client user. Calling Number Identification received across the PRI will also be forwarded to the VMS. The VMS will play the client user's personalized greeting and the caller's voice message will be stored. If the call originated from a phone within the same administrative environment, the calling number will also be stored. When reviewing their messages, the client user can respond by sending a message with voice messaging using VMS-to-VMS messaging capabilities with the stored calling number as the destination address.

Obtain Equipment with These Capabilities

Voice Messaging System (VMS)

- PBX/VMS Integration link.
- Networking capabilities for mail box-to-mail box message sending.

Private Branch Exchange (PBX)

- PBX/VMS Integration link.
- PRI, with voice calling on each B channel.

Request These Capabilities From ISDN Service Provider

Network Interface to Private Branch Exchange (PBX)

- One 23B+D PRI.
- Directory Numbers (DNs) as required by application.
- B channels with:

  Voice

- Features for Voice:
  - Calling Number Identification Services

Technical Notes

- Messages can be sent between systems as data, using, for example, Simple Message Transfer Protocol (SMTP) on a circuit-data call over the PRI.
- Administration can be centralized using the VMS vendor's application protocol on a circuit-data call over the PRI.
- For detailed Signaling Flow Diagrams for this application, see Sections 5.2.3.3 (Voice Call Setup), and 5.2.3.8 (Voice & Circuit-Data Call Clearing).