EXHIBIT B

(FCC Ref. 2.1033(b)(4))

"Description of Circuit Functions"

TT150 SYSTEM DESCRIPTION

1.0 Connectors

Microphone Input
Headset Output
RJ11 for direct connection of telephone set
RJ11 for connection to central office
Footswitch Input

2.0 Functions

Record and Playback using Mic/Headset, Direct Connect telephone, or Telephone Line.
Transcription using Headset/footswitch

3.0 Subsystem Functional blocks

3.1 Direct connect phone
3.2 Telco Interface
3.3 DSP voice engine
3.4 OKI voice engine
3.5 Telephone Signalling
3.6 Audio Control

3.1 Direct connect phone

The Direct Connect Phone connection is an RJ11,J1 powered by +9V and -9V through 2 resistors R9 & R19. An opto coupler U9 is used to detect current through the phone loop to signal an Off Hook condition.

When an Off Hook condition is recognized by software, Relay RL2 is energized and switches the DAA 2 wire side to connect to J1 in parallel. This sets up a powered 2 wired audio path between the Direct Connect Phone and the DAA. RL2 ensures that the Direct Connect Phone can never connect to the telco line.

3.2 Telco Interface

The Telco interface consists of an RJ11, J2, Protection circuitry Polyswitch F1 and Sidactor CR4, current detector U21, Relay RL2 and DAA U8.

Relay RL2's Normally Closed contacts connect the Telco to the DAA. The DAA does Ring Detection, Loop Seizure, 2 to 4 wire conversion and isolation.

When a ring is detected, the software sends a loop seizure signal to the DAA to seize the line. This sets up an audio path from the Telco to the selected Voice Engine and the Telephone signalling decoders.

U21 and opto coupler detects the presence/absence of loop current supplied by the Central office.

3.3 DSP voice engine

The DSP voice engine consists of Codec U24, the dedicated True Speech processor U16, and the ISA interface PLD's U28 and U29. The Audio from the DAA is mixed with the audio from the microphone and then processed an Automatic Voice Level Control Consisting of U6 and U7.

The level controlled audio is then digitized by the Codec and sent to the DSP. The DSP processes the digitized audio and sends it to the ISA bus.

For playback, the codec reconstructs the analog which is attenuated by volume control U12 switched by U14 a 4066 to the DAA mixer U25C. A separately attenuated signal from the volume control goes to U32 the headset amplifier.

3.4 OKI voice engine

The Oki voice engine consists of U15, and Oki Codec, and U26 an Intel Microcontroller. The U15 digitizes the audio and presents the digital word in 4 bit chunks to the Microcontroller. The microcontroller packages the chunks into 8 bit Bytes and sends them to the ISA bus.

For playback, the OKI reconstructs the analog which is attenuated by volume control U12 switched by U14 a 4066 to the DAA mixer U25C. A separately attenuated signal from the volume control goes to U32 the headset amplifier.

3.5 Telephone Signalling

The Audio from the 4 wire side of the DAA goes to U13 a 1200 baud FSK demodulator and U19 and U20 a pair of DTMF and call progress chips.

When microcontroller U27 detects the first ring, it processes the data from U13, the Caller Id decoder. It then signals the ISA bus that it has valid data if the CID data was processesed successfully.

DTMF chips U19 and U20 can be setup to detect DTMF or Call Progress signalls.When a valid signal is detected the internal registers are updated and a readable flag is set. U20 is also capable of generating DTMF tones. In this mode, the tone goes to the DAA mixer, is enabled by the 4066 switch and sent to the DAA.

3.6 Audio Control

The microphone signal is amplified by U22 and then mixed with the DAA signal in mixer U25A and then goes to Automatic level control U6.

The outputs of the Codecs go to Audio Selector U12. U12 selects the audio source, contours the Bass and Treble and attenuates the audio before sending it to the DAA mixer and the Headset Amplifier.