Network Convergence of Voice and Data Technology Test for the Efficiency of Listening Quality Using VoDSL

Sarhan M. Musa[1]

Abstract -**This paper presents a baseline test for the listening quality (LQ) using voice over digital subscriber line (VoDSL) access technology by using voice/listening quality(V/LQ) transmission with voice compression while countinously downloading file. The design of an experimental VoDSL network architecture is presented. We identify the efficiency of the LQ based on the following digital subscriber line (DSL) service levels, 640K/640K, 1.5M/256K, and 3.0M/512K for each American National Standards Institute (ANSI) loops.**

Keywords **-Voice over Digital Subscriber Line (VoDSL), Digital Subscriber Line (DSL), Integrated Access Devices (IAD), Listening Quality (LQ).**

I. INTRODUCTION

The next generations of telecommunications providers around the globe are engaging in the rapid development of a new service that will combine both data communications and telephony. VoDSL has been developed with the rapid increase of the Internet and of the data traffic through network convergence of voice and data [1]. VoDSL service has the capability to provide the customers with converged voice and data, including local and long distance telephone service, plus high speed Internet access, on a single DSL copper line. The tests and the requirements demonstrated of VoDSL equipment for voice functionality and voice quality (VQ) in the DSL forum technical report are in [2]. We used on our test asymmetric digital subscriber lines (ADSL), because it provides a "life-line" capability, so that if the power fails, one telephone line will still work. It has a lower bandwidth upstream than downstream. It transmits high bit rate data in the down direction from the central office (CO) to the subscriber (downstream), with typical bit rates from 1.5 to 8 Mb/s, and lower bit rate data in the reverse (up) direction from the subscriber to the CO (upstream), with bit rates from 64 to 640 Kb/s [3]. ADSL is used for asymmetric services to residential and small office home office (SOHO) customers. This paper contains results that can be used in evaluating VoDSL solutions that offers multiple voice connections simultaneously with data onto the high speed digital line offered by ADSL line. The results are based on the V/LQ of VoDSL. In section 2, we present description of the network architecture for V/LQ test setup.

In section 3, we have a discussion on the subscriber loop plant noise. Section 4 is the result of the tests performed while section 5 is the conclusion.

[1]Sarhan M. Musa is with the Faculty of Engineering Technology, Prairie A&M University, Prairie View, TX 77446, USA, E-mail:sarhan_musa@pvamu.edu

II. DESCRIPTION OF THE NETWORK ARCHITECTURE

We used the V/LQ transmission with voice compression while continuously downloading file to verify the ability to support up to eight derived lines and the LQ on the integrated access devices (IADs) for VoDSL. The following DSL service levels were used 640K/640K, 1.5M/256K and 3.0M/512K for each ANSI loops. Figure 1 shows the fixed wireline networks test setup for the access evolution of broadband services using VoDSL technology for V/LQ transmission. We verified the V/LQ transmission with voice compression and the maximum number of line connections in compressed mode operation that can be supported without any problems based on figure 1.

The customers can have multiple IADs based on their needs. Each IAD has 4-8 telephone interfaces plus an Ethernet interface. We test one IAD based on VoDSL solutions. Eight telephones are connected to the IAD that resides in the customer premises through plain old telephone service (POTS). Data from the personal computer (PC) source running file transfer protocol (FTP) is provided to the IAD via the Ethernet. IAD interconnects the customer's premises equipment (CPE) and ADSL service and it converts POTS to Asynchronous transfer mode adaptation layer type 2 (AAL2). It uses the same virtual path identifier/virtual channel identifier (VPI/VCI) for voice and data. So these have to be mapped to unique values before transmission on the same pipe out of the digital subscriber line access multiplexer (DSLAM), for example, 0/39 to 0/209. All the Voice calls are digitized in the form of AAL2 cells and sent over the same permanent virtual circuit (PVC), for example, 0/38. AAL2 allows us to use the same VPI/VCI for multiple users. Users do not necessarily have to coexist in an asynchronous transfer mode (ATM) cell. Each cell can carry data from just one user. The data rides separate PVC (e.g. 0/39). Both these PVCs are transported over the ADSL copper connection to a DSLAM. The DSLAM aggregates all the PVCs onto a single connection, for example, a DS-3, and sends it to an ATM switch. The ATM switch separates the voice and data calls. The voice PVCs go to a voice gateway. The voice gateway converts the ATM traffic into time division multiplexing (TDM) analog traffic and interfaces to a Class 5 switch via GR-303 Interfaces. The voice calls go to public switched telephone network (PSTN) that connected to POTS, which carries the voice calls to the telephones. On the other hand, the data is sent to a router through digital signal-level 3 (DS-3) and to the Internet then to a PC running FTP through Ethernet.

We need to confirm the dial tone on all the ports of the IAD by connecting phones to the ports and taking them off hook at the same time as well as individually and recording the number of phones that simultaneously have dial tone. We make calls from each port of the IAD to a phone connected to the PSTN, and we repeat the same from a PSTN phone to each port of the IAD. We enable voice compression and verify the number of phones having dial tone. These tests performance are based on the Asymmetric Digital Subscriber Line Wireline Simulator (ADSL WLS) which is located between the IAD (at customer) and DSLAM (at CO) is used to generate and receive the traffic between the customer and CO after assigning the following loops: ANSI Loop #7 with 24 DSL disturbers, ANSI Loop #9 with 24 DSL disturbers, and ANSI Loop #13 with 24 DSL disturbers. Figure 2 shows some examples of the copper loops impairments as made up of pairs from sections of several cables between the central office and the customer.

III. SUBSCRIBER LOOP PLANT

The connections of the telephones by the twisted pair of copper wires to the network can experience some form of noise. The noise arises from the thermal noise of the twisted pair itself, the noise generated internally by the receiving modem, and signals electromagnetically coupled into the phone line [4]. We will concentrate on crosstalk noise, that is, the undesired coupling of a signal from one communication channel to another, and it occurs when some of the transmissions signal energy leaks from the cable. There are two types of crosstalk: near-end crosstalk (NEXT) and far-end crosstalk (FEXT). NEXT is the result of a leaking from nearby transmitting source into a receiver through the coupling between pairs. FEXT is the noise detected by the receiver located at the far end of the cable from the transmitter (noise source). Figure 3 shows two types of DSL crosstalk. Where pairs j and i belong to the same distribution cable and operate in full duplex. NEXT and FEXT models are specified in appropriate DSL standards for the purpose of guiding simulation study.

Galli and Kerpez [5] studied the theoretical analysis methods of summing crosstalk mixed sources. According to [5] the received power spectral densities (PDSs) of NEXT and FEXT due to more than one crosstalk disturber for n interfering signals of the same kind become

$$N_{ext}[f,n] = S(f)X_N f^{1.5} n^{0.6} \quad (1)$$

$$F_{ext}[f,n,l] = S(f)X_F f^2 l\,|H(f)|^2\, n^{0.6} \quad (2)$$

where n is the number of interfering signals, f is the frequency, $S(f)$ is the PSD of the disturbing signal, l is the loop length , $H(f)$ is the transfer function of the loop, and X_N and X_F are constants determined by measurements. Now, when $n=1$, that is, one pair-to-pair crosstalk disturber, thus

$$N_{ext}[f] = S(f)X_N f^{1.5} \quad (3)$$

$$F_{ext}[f,l] = S(f)X_F f^2 l\,|H(f)|^2 \quad (4)$$

the starting point is expressed in the 1 % worst case for crosstalk. Crosstalk noise at high frequency can be a major limitation for providing high speed digital communications through the twisted pair loop plant. However, it can be ignored at voice frequency because it is very small. According to Werner [6] the signal-to-noise ratio (SNR) under NEXT can be expressed as

$$SNR_n \approx \frac{e^{-2d\zeta\sqrt{f}}}{\chi f^{1.5}} \quad (5)$$

where d is the loop distance in feet, ζ is constant equal to 9 x 10^{-7} for 2 gauge loop, χ is constant equal to 8.8 x 10^{-14} for the 49 disturber 1 % worst case NEXT model, and f is the frequency in Hz. The SNR under FEXT can be expressed as

$$SNR_f = \frac{1}{\psi f^2 d} \quad (6)$$

where ψ is constant equal to 8 x 10^{-20} for the 49 disturber 1% worst case FEXT model. The SNR bandwidth limited by a receiver background white noise (AWGN) can be expressed as

$$SNR_w = 1 \times 10^{10} e^{-2d\zeta\sqrt{f}} \quad (7)$$

where assuming that a -40dbm/Hz transmitted power density level and a -14 dbm/Hz receiver background noise power density level.

IV. RESULTS

We used analog phones for testing, and the V/LQ is done using a digital speech level analyzer (DSLA) from Malden Electronics. This generates phonetically balanced speech samples and assesses the Mean Opinion Score (MOS) value which is claimed to have very good correlation with actual subjective assessment. The scaling for listening quality based on five categories: 4 +x (Excellent), 3 +x (Good), 2 +x (Fair), 1+x (Poor), 0 or other (Unacceptable), where x is a decimal number. The following results in figures 4-12 were observed for the DSL service levels 640K/640K, 1.5M/256K, and 3.0M/512K for each ANSI loops mentioned previously: First, the LQ_s are excellent at both speeds for the customer and the variations are very small in range of 1% for each DSL level. However, if we need to be more concerned on the 1% range, it is clearly shown that the LQ for -10 dbm (excellent quality) is greater than -40dbm (good quality) for both DSL service levels. Second, the LQ is better for the customer when the perceptual analysis measurement systems (PAMS) from the CO to the customer than the verse Third, the highest LQ for the customer is 4.31 for ANSI loop #13 with 24 DSL disturbers at speeds 640k / 640k. The lowest LQ for the customer is 4.27 for ANSI Loop #7 with 24 DSL disturbers at the same speed. But, the highest LQ for the customer is 4.27 for ANSI loop #9 and loop #13 with 24 DSL disturbers at speeds 1.5 KM/256K. The lowest LQ for the customer is 4.26 for ANSI Loop #7 with 24 DSL disturbers at 1.5M/ 256k. On the other hand, the highest LQ for the customer is 4.29 for ANSI loop #9 with 24 DSL disturbers at speed 3.0M / 512k. The lowest LQ for the customer is 4.26 for ANSI Loop #13 with 24 DSL disturbers at 3.0M / 512k.

V. CONCLUSION

It has been shown that the following DSL service levels 640K/640K,1.5M/256K and 3.0M/512K, for each American National Standards Institute (ANSI) loops have been used to provide better voice/listening quality for customers. From the previous figures 4-12, it is clearly shown that -40dbm has more attenuation (cable loss) than -10dbm, which made the last produce better LQ using VoDSL. The downstream band is greater than the upstream that can make the LQ better for customer who receives the call from the CO than verse. The listening qualities are in excellent service for customer and the variation is very small limited to 1% range for the DSL service levels 640K/640K, 1.5M/256k and 3.0M/512k for each ANSI loop.

REFERENCES

[1] R. Franz, "Optimal Migration towards Voice/Data Convergence," *ICCT 2000 contribution.*

[2] DSL Forum Technical Report TR-049, "*VoDSL Interoperability Test plan*," May 2002.

[3] Network and Customer Installation Interfaces-Asymmetric Digital Subscriber Line (ADSL) Metallic Interface, ANSI Standard T1.413, *1995*.

[4] J. Cioffi, P. Silverman, and T. Starr, "Digital subscriber lines," *Computer Networks 31, 283-311 (1999)*

[5] Stefano Galli and Kenneth J. Kerpez, "Methods of summing crosstalk from mixed sources-part I: theoretical analysis," *IEEE Transaction on Communications, Vol. 50, No. 3, march 2002.*

[6] J. J. Werner, "The HDSL environment," *IEEE Transaction selected areas in communications. 9 (6) (1991) 785-800.*

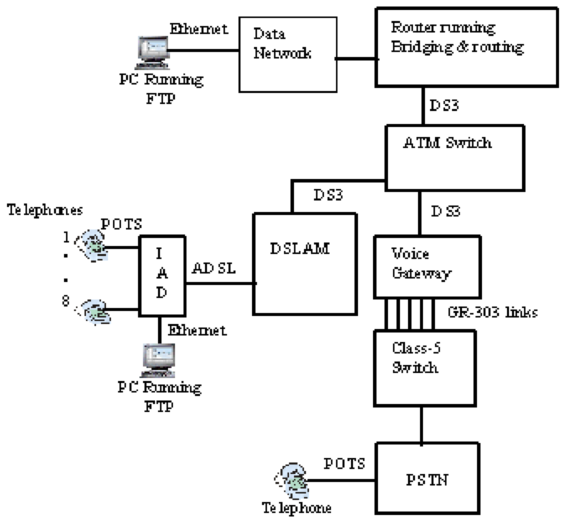

Fig. 1. Voice quality test transmission configuration

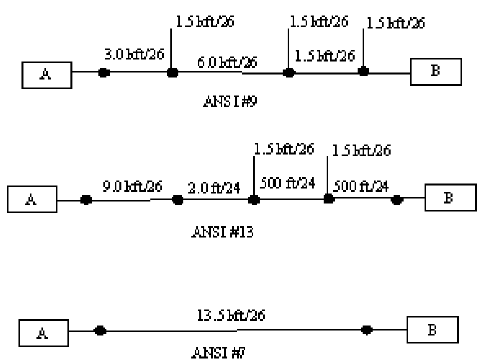

Fig. 2. Examples of ANSI loop descriptions

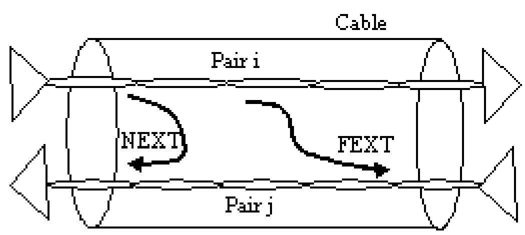

Fig. 3. NEXT and FEXT crosstalk

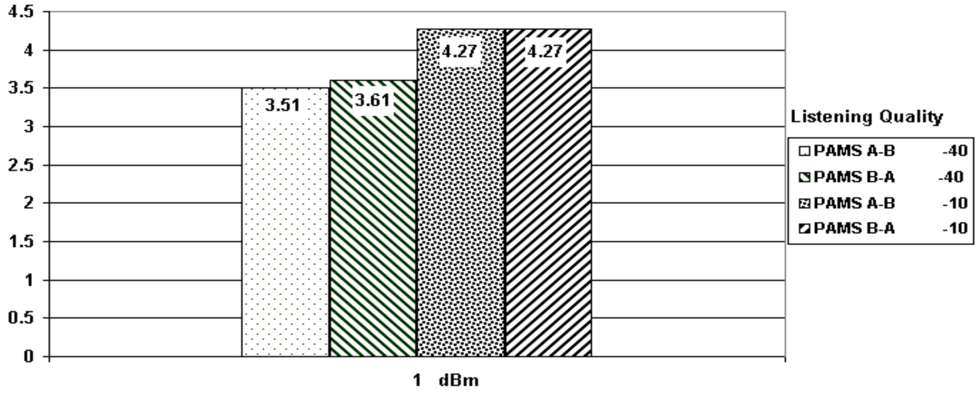

Fig. 4. DSL Service Level 640k / 640k for ANSI Loop #7 with 24 DSL disturbers

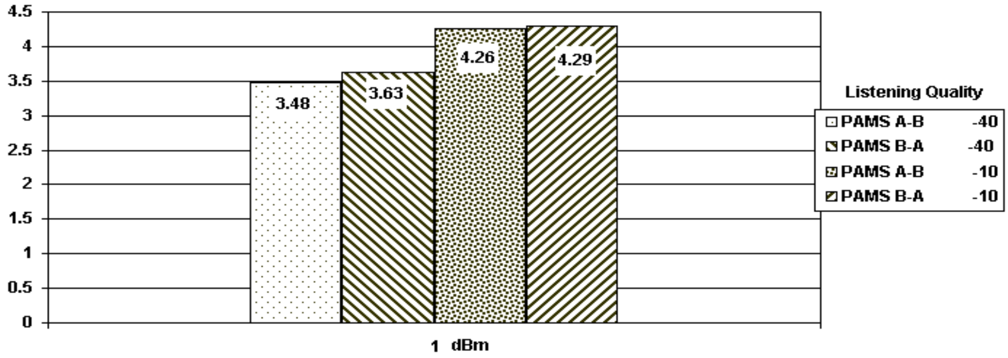

Fig. 5. DSL Service Level 640k / 640k for ANSI Loop #9 with 24 DSL disturbers

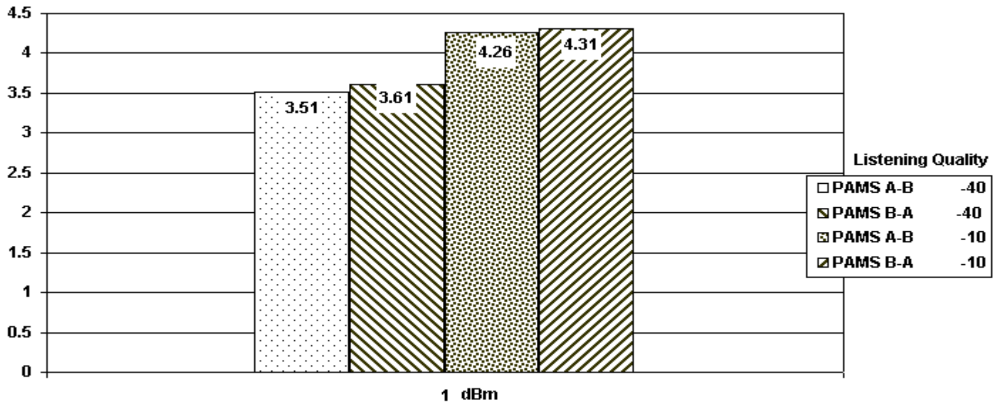

Fig. 6. DSL Service Level 640k / 640k for ANSI Loop #13 with 24 DSL disturbers

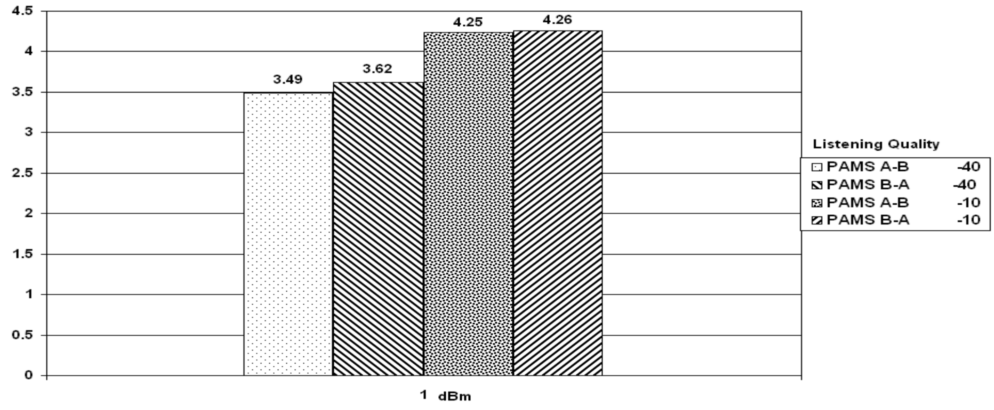

Fig. 7. DSL Service Level 1.5 KM/256K for ANSI Loop #7 with 24 DSL disturbers

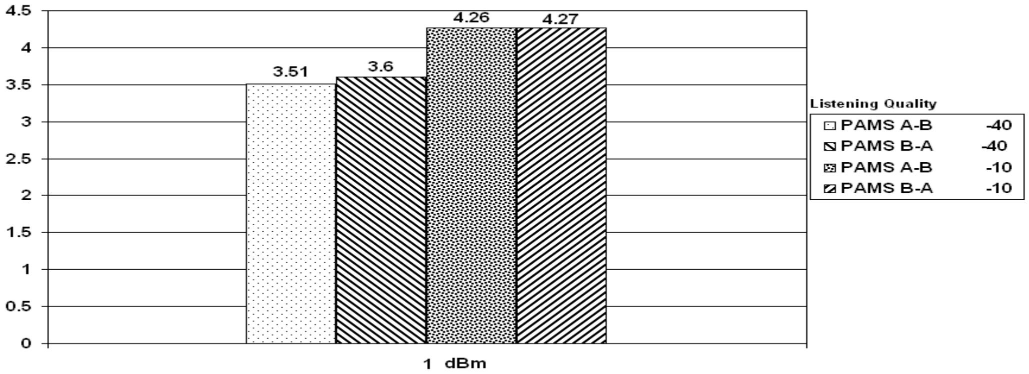

Fig. 8. DSL Service Level 1.5M / 256k for ANSI Loop #9 with 24 DSL disturbers

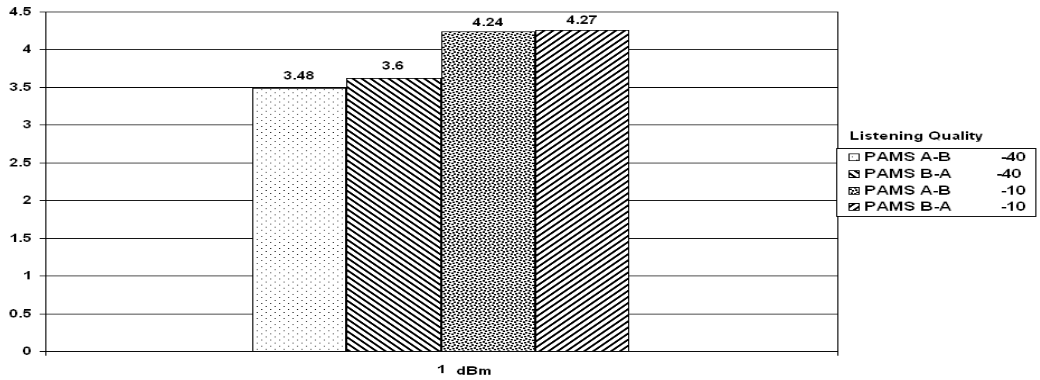

Fig. 9. DSL Service Level 1.5M / 256k for ANSI Loop #13 with 24 DSL disturbers

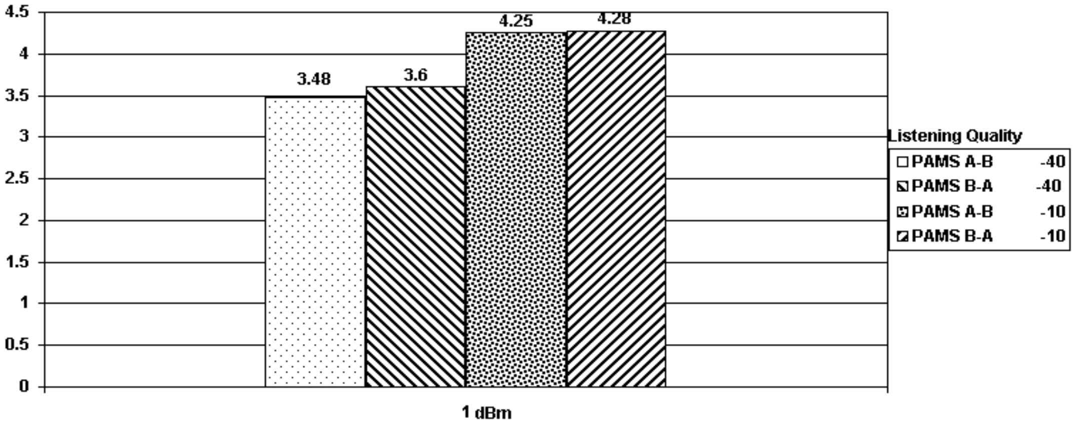

Fig. 10. DSL Service Level 3.0M / 512k for ANSI Loop #7 with 24 DSL disturbers

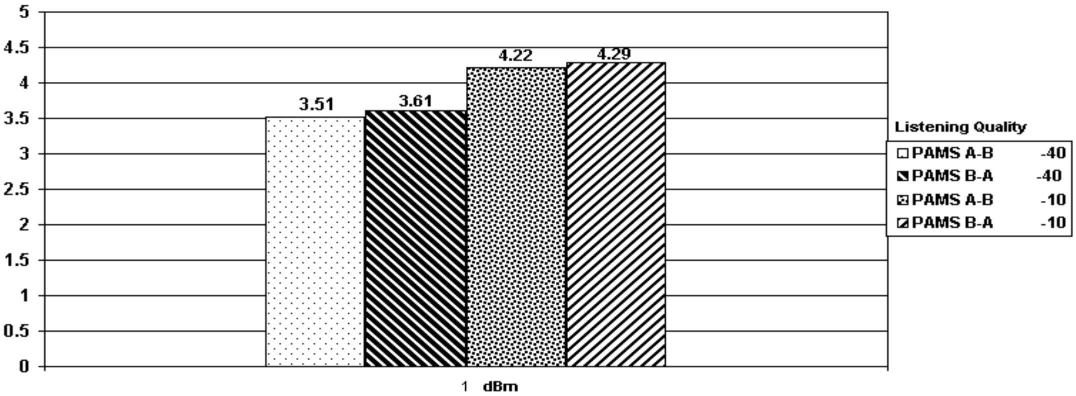

Fig. 11. DSL Service Level 3.0M / 512k for ANSI Loop #9 with 24 DSL disturbers

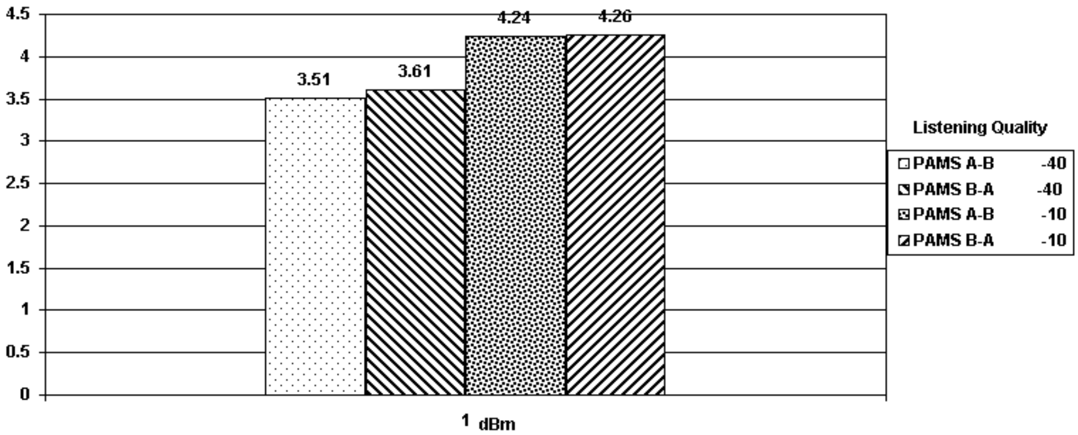

Fig. 12. DSL Service Level 3.0M / 512k for ANSI Loop #13 with 24 DSL disturbers